DEFENSE INFORMATION SYSTEMS AGENCY
P. O. BOX 549
FORT MEADE, MARYLAND 20755-0549

IN REPLY REFER TO: Joint Interoperability Test Command (JTE) 20 Apr 16

MEMORANDUM FOR DISTRIBUTION Revision 1

SUBJECT: Joint Interoperability Certification of the GENBAND Inc. EXPERiUS Release 11.2

References: (a) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(b) Office of the Department of Defense Chief Information Officer, "Department of Defense Unified Capabilities Requirements 2013, Errata 1," 1 July 2013
(c) through (d), see Enclosure 1

1. **Certification Authority.** Reference (a) establishes the Joint Interoperability Test Command (JITC) as the Joint Interoperability Certification Authority (CA) for UC products.

2. **Conditions of Certification.** The GENBAND Inc. EXPERiUS Release 11.2; hereinafter referred to as the System Under Test (SUT), meets the critical requirements of the Unified Capabilities Requirements (UCR), Reference (b), and is certified for joint use as a Local Session Controller (LSC) with the conditions described in Table 1. This certification expires upon changes that affect interoperability, but no later than three years from the date of the UC Approved Products List (APL) memorandum.

Table 1. Conditions

Condition	Operational Impact	Remarks
UCR Waivers		
Per the vendor's LoC, the SUT does not fully support IPv6. The DoD CIO waived all of the IPv6 discrepancies noted in this certification letter on 28 July 2015 with vendor's POA&M. Therefore, IPv6 was not tested and is not included in this certification.		
Conditions of Fielding		
The SUT is certified in the United States, including the CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.		
During testing, the AudioCodes MG3K and M800 could not dynamically invoke VBD G.711 or V.150.1. The SUT supports V.150.1 however when G.711 Voice Codec is negotiated the SUT fails to support bi-directional secure calls in pass-through mode. The SUT DISA has accepted the vendor's POA&M and adjudicated this as minor with the Condition of Fielding that the SUT cannot operate with V.150 until this discrepancy is corrected.		
Open Test Discrepancies		
Per the vendor's LoC, TEO AEIs do not support integration with directory.	Minor	See note 1.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies (continued)		
During testing, the SUT did not allow IP users to place emergency calls without authenticating TLS.	None	See note 2.
Per the vendor's LoC, the SUT did not fully support SNMPv3.	None	See note 2.
Per the vendor's LoC, the SUT does not fully support RFC 4213.	None	See note 3.
Per the vendor's LoC, the SUT does not fully support RFC 4291.	None	See note 3.
Per the vendor's LoC, the SUT does not support RFC 4007.	None	See note 3.
Per the vendor's LoC, the SUT does not support RFC 4861.	None	See note 3.
Per the vendor's LoC, the SUT's MAS does not support IPv6.	None	See note 3.
Per the vendor's LoC, the Polycom VVX ROEI does not support ANAT.	None	See note 3.
Per the vendor's LoC, the SUT does not support RFC 3484.	None	See note 3.
The Polycom 310, 410, 510, and 610 EIs do not divert precedence above ROUTINE calls to an attendant or Alternate DN.	Minor	See note 4.
Per the vendor's LoC, the SUT soft client does not support IPsec and RFC 4301.	None	See notes 3, 4.
During testing, the SUT soft client was unable to establish two-way video with other video systems.	Minor	See note 5.
During testing, the SUT allowed unlike service domains to preempt.	Minor	See note 6.
During testing, the SUT sent a 500-server error when an unanswered trunk call was preempted for reuse.	Minor	See note 6.
During testing, the SUT AudioCodes gateway did not support Method 2 hunt sequence.	Minor	See note 7.
During testing, the SUT did not support dial pulse signaling.	Minor	See note 7.
During testing, the SUT did not support Domain Directory per the requirement.	Minor	See note 6.
During testing, the Polycom VVX 500 and VVX600 video phones have one-way video with Cisco 99xx video EI.	Minor	See note 6.
During testing, the SUT did not support OPTIONS Requests.	Minor	See note 6.

NOTES:
1. DISA has adjudicated this discrepancy as minor.
2. All IA requirements in UCR, section 4, have been changed in UCR 2013, Change 1, with the following caveat: The requirements in section 4 will not be evaluated in interoperability test plans and are the responsibility of cyber security testing with the intent to minimize redundancy in cyber security test procedures and reports. The update in UCR 2013, Change 1 has immediate applicability.
3. The DoD CIO waived all of the IPv6 discrepancies noted in this certification letter on 28 July 2015 with vendor's POA&M. Therefore, IPv6 was not tested and is not included in this certification.
4. DISA has adjudicated this discrepancy as minor. This requirement was changed in the UCR 2013, Change 1 to allow precedence calls above ROUTINE to be answered by an ROEI if resources are available.
5. DISA has accepted the vendor's POA&M and adjudicated this as critical for video on the soft client. The soft client is certified for audio only.
6. DISA has accepted the vendor's POA&M and adjudicated this discrepancy as minor.
7. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to conditional in the next version of the UCR.

LEGEND:

AEI	AS-SIP End Instrument
ANAT	Alternative Network Address Types
AS-SIP	Assured Services Session Initiation Protocol
CIO	Chief Information Officer
CONUS	Continental United States
DISA	Defense Information System Agency
DN	Directory Number
DoD	Department of Defense
E1	European Basic Multiplex Rate
EI	End Instrument
ETSI	European Telecommunications Standards Institute
IA	Information Assurance
IP	Internet Protocol
IPsec	Internet Protocol Security
IPv6	Internet Protocol version 6
LoC	Letter of Compliance
MAS	Media Application Server
POA&M	Plan of Action and Milestones
RFC	Request For Comment
ROEI	Routine Only End Instrument
SNMPv3	Simple Network Management Protocol version 3
SUT	System Under Test
TLS	Transport Layer Security
UCR	Unified Capabilities Requirements
VBD	Voice Band Data
VVX	Voice and Video eXchange

3. **Interoperability Status.** Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. Table 4 provides the UC APL product summary.

Table 2. Interface Status

Interface	Applicability (See note 1.)	Status	Remarks
Network Management Interfaces			
10BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface.
100BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface.
1000BaseT	Conditional	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface.
Network Interfaces (Line and Trunk)			
10BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface with the SUT PEIs and softphones.
100BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface with the SUT PEIs and softphones.
1000BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface with the SUT PEIs and softphones.
2-wire analog (line only)	Required	Met	The SUT met the critical CRs and FRs for the 2-wire analog interface with the SUT 2-wire secure and non-secure analog instruments.
ISDN BRI	Conditional	Not Tested	The SUT does not support this conditional line interface.
Legacy Interfaces (External)			
ISDN T1 PRI (ANSI T1.619a)	Required	Met	The SUT met the critical CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Required	Met	The SUT met the critical CRs/FRs. This interface provides PSTN connectivity.
T1 CCS7 (ANSI T1.619a)	Conditional	Not Tested	The SUT does not support this conditional interface.
T1 CAS	Conditional	Not Tested	Although the SUT supports this conditional interface, it was not tested and is not covered under this certification.
E1 PRI (ITU-T Q.955.3)	Required	Not Tested	The SUT does not support this required interface. This interface provides OCONUS MLPP connectivity in ETSI-compliant countries. (See note 2.)
E1 PRI (ITU-T Q.931)	Required	Not Tested	The SUT does not support this required interface. This interface provides OCONUS connectivity in ETSI-compliant countries. (See note 2.)

NOTES:
1. The SUT high-level requirements are depicted in Table 3. These high-level requirements refer to a detailed list of requirements provided in Enclosure 3.
2. The SUT is certified in the United States, including the CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.

LEGEND:

10BaseT	10 Mbps Ethernet
100BaseT	100 Mbps Ethernet
1000BaseT	1000 Mbps Ethernet
ANSI	American National Standards Institute
AS-SIP	Assured Services Session Initiation Protocol
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
CCS7	Common Channel Signaling Number 7
CONUS	Continental United States
CR	Capability Requirement
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
ETSI	European Telecommunications Standards Institute
FR	Functional Requirement
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
Mbps	Megabits per second
MLPP	Multi-Level Precedence and Preemption
NI-2	National ISDN Standard 2
OCONUS	Outside the Continental United States
PEI	Proprietary End Instrument
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
Q.955.3	ISDN Signaling Standard for E1 MLPP
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1

Table 3. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	Voice Features and Capabilities (R)	2.2	Met
2	Assured Services Admission Control (R)	2.3	Met
3	Signaling Protocols (R)	2.4	Partially Met (See note 2.)
4	Registration and Authentication (R)	2.5	Met
5	SC and SS Failover and Recovery (R)	2.6	Met
6	Product Interface (R)	2.7	Met
7	Product Physical, Quality, and Environmental Factors (R)	2.8	Met
8	End Instruments (including tones and announcements) (R)	2.9	Partially Met (See note 2.)
9	Session Controller (R)	2.10	Met
10	AS-SIP Gateways (C)	2.11	Met
11	Call Connection Agent (R)	2.14	Met
12	CCA Interaction with Network Appliances and Functions (R)	2.15	Met
13	Media Gateway (R)	2.16	Partially Met (See note 2.)
14	Worldwide Numbering & Dialing Plan (R)	2.18	Partially Met (See note 2.)
15	Management of Network Devices (R)	2.19	Met
16	V.150.1 Modem Relay Secure Phone Support (R)	2.20	Partially Met (See note 2.)
17	Requirements for Supporting AS-SIP Based Ethernet Devices for Voicemail Systems (C)	2.21	Met
18	Local Attendant Console Features (O)	2.22	Not Tested
19	MSC and SSC (O)	2.23	Not Tested
20	MSC, SSC, and Dynamic ASAC Requirements in Support of Bandwidth-constrained links (O)	2.24	Not Tested
21	Other UC Voice (R)	2.25	Partially Met (See note 2.)
22	Information Assurance Requirements (R)	4	Partially Met (See notes 2, 3.)
23	IPv6 Requirements (R)	5	Not Tested (See note 4.)
24	Assured-Services (AS) Session Initiation Protocol (SIP) (AS-SIP 2013) (R)	AS-SIP	Partially Met (See note 2.)

NOTES:
1. The annotation of ‘required’ refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.
2. The SUT met the requirements with the exceptions noted in Table 1. DISA adjudicated these exceptions as minor.
3. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (d).
4. The DoD CIO waived all of the IPv6 discrepancies noted in this certification letter on 28 July 2015 with vendor’s POA&M. Therefore,, IPv6 was not tested and is not included in this certification.

LEGEND:

ASAC	Assured Services Admission Control	MSC	Master Session Controller
AS-SIP	Assured Services Session Initiation Protocol	O	Optional
C	Conditional	POA&M	Plan of Action and Milestones
CCA	Call Connection Agent	R	Required
CIO	Chief Information Officer	SC	Session Controller
CR	Capability Requirement	SS	Softswitch
DISA	Defense Information System Agency	SSC	Subtended Session Controller
DoD	Department of Defense	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements
IPv6	Internet Protocol version 6		

Table 4. UC APL Product Summary

Product Identification			
Product Name	GENBAND Inc. EXPERiUS		
Software Release	Release 11.2		
UC Product Type(s)	Session Controller		
Product Description	LSC consists of: EXPERiUS Application Server (Personal Agent, Session Manager, Media Application Server), Management Workstation, AudioCodes Mediant 3000, GENCom for Windows Soft Client, Polycom VVX, TEO AEI.		
Product Components (See note 1.)	**Component Name (See note 2.)**	**Version**	**Remarks**
Application Server	**EXPERiUS Application Server** (Personal Agent, Session Manager, Media Application Server)	VMWare vSphere Hypervisor ESXi 5 RHEL v6.6 Linux 2.6.32-504.30.3.el6x86_64 Personal Agent (VMWare) (x2) MCP 17.0 Session Manager (VMWare) (x2) MCP 17.0 Media Application Server (MAS) (X2) MAS 16.0	
TDM to IP Gateway	**AudioCodes Mediant 3000 (M3000) Gateway**	Mediant 6.60A.305.001 pSoS 2.5.4	
TDM to IP Gateway	**AudioCodes M800 Gateway**	Mediant 6.60A.305.001 Embedded Linux Kernel 2.6.21.7	
IP Phone (Voice)	TEO **7810**, **4104** 4101 AS-SIP and SIP telephones	xx.04.22	(See note 3.)
IP Phone (Voice)	**Polycom VVX 310, VVX 410**	5.4.2.0334	
IP Phone (Voice and Video)	**Polycom VVX 500 and VVX 600**	5.4.2.0334	
Management Workstation (Site-provided Dell Laptop)	**Dell Laptop**	Windows 7 SP 1 Axway Desktop Validator 4.11.2.753 ActivClient 6.2.0.50	
Soft Clients (Site-provided Dell Laptop)	**GENBAND Multimedia Client**	Windows 7 SP 1 GENCom 10.4 v 10.4.1368 Axway Desktop Validator 4.11.2.753 ActivClient 6.2.0.50	Certified for voice only.

NOTES:
1. The detailed component and subcomponent list is provided in Enclosure 3.
2. Components bolded and underlined were tested by JITC. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes.
3. The TEO IP 7810, 4104, and 4101 phone units were tested using version xx.04.22 from the previous xx.04.21.

LEGEND:

AEI	AS-SIP End Instrument	MAS	Media Application Server
APL	Approved Products List	RHEL	Red Hat Enterprise Linux
AS-SIP	Assured Services Session Initiation Protocol	SIP	Session Initiation Protocol
DISA	Defense Information System Agency	SP	Service Pack
DSN	Defense Switched Network	TDM	Time-Division Multiplexing
IP	Internet Protocol	UC	Unified Capabilities
JITC	Joint Interoperability Test Command	VVX	Voice and Video eXchange
LSC	Local Session Controller		

4. **Test Details.** This certification is based on interoperability testing, review of the vendor's Letters of Compliance (LoC), Defense Information System Agency (DISA) adjudication of open Test Discrepancy Reports (TDRs), and DISA CA Recommendation for inclusion on the UC APL. Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 6 July 2015 through 11 February 2016 using test procedures derived

from Reference (c). Review of the vendor’s LoC was completed on 10 February 2016. DISA adjudication of outstanding TDRs was completed on 2 March 2016. Patches were applied and configuration changes were made. Verification and Validation testing was conducted from 21 through 25 March 2016. Information Assurance testing was conducted by DISA-led Information Assurance test teams and the results are published in a separate report, Reference (d). Enclosure 2 documents the test results and describes the tested network and system configurations. Enclosure 3 provides a detailed list of the interface, capability, and functional requirements. Enclosure 4 provides a list of errata changes to this certification since the original signature date.

5. **Additional Information.** JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at https://stp.fhu.disa.mil/. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil/. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly from the Unified Capabilities Certification Office (UCCO), e-mail: disa.meade.ns.list.unified-capabilities-certification-office@mail.mil. All associated information is available on the DISA UCCO website located at http://www.disa.mil/Services/Network-Services/UCCO.

6. **Point of Contact (POC).** The JITC point of contact is Capt. Soamva Duong, commercial telephone (520) 538-5269, DSN telephone 879-5269, FAX DSN 879-4347; e-mail address soamva.duong.fm@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Capt. Soamva Duong) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 150301.

FOR THE COMMANDER:

Bradley A. Clark

4 Enclosures a/s

for RIC HARRISON
Chief
Networks/Communications and UC Division

Distribution (electronic mail):
DoD CIO
Joint Staff J-6, JCS
USD (AT&L)
ISG Secretariat, DISA, JTA
U.S. Strategic Command, J665
US Navy, OPNAV N2/N6FP12
US Army, DA-OSA, CIO/G-6 ASA(ALT), SAIS-IOQ
US Air Force, A3CNN/A6CNN
US Marine Corps, MARCORSYSCOM, SIAT, A&CE Division
US Coast Guard, CG-64
DISA/TEMC
DIA, Office of the Acquisition Executive
NSG Interoperability Assessment Team
DOT&E, Netcentric Systems and Naval Warfare
Medical Health Systems, JMIS IV&V
HQUSAISEC, AMSEL-IE-IS
UCCO

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, "Session Controller (SC) Test Procedures Version 1.0 for Unified Capabilities Requirements (UCR) 2013 Errata 1," June 2014
(d) Joint Interoperability Test Command, "Information Assurance (IA) Findings Summary For Genband Inc. EXPERiUS Release (Rel) 11.2 (Tracking Number 1510301)," Draft

CERTIFICATION SUMMARY

1. SYSTEM AND REQUIREMENTS IDENTIFICATION. The GENBAND Inc. EXPERiUS Release 11.2 is hereinafter referred to as the System Under Test (SUT). Table 2-1 depicts the SUT identifying information and requirements source.

Table 2-1. System and Requirements Identification

System Identification	
Sponsor	Defense Information Systems Agency
Sponsor Point of Contact	Wooten, Stanley, 6916 Copper Ave, Ft. Meade, MD, 20755-0549 email: Stanley.e.wooten.civ@mail.mil
Vendor Point of Contact	Travis, Richard, 3605 East Plano Parkway, Plano, TX 75074, email: richard.travis@genband.com
System Name	GENBAND EXPERiUS
Increment and/or Version	Release 11.2
Product Category	Local Session Controller (LSC)
System Background	
Previous certifications	None
Tracking	
UCCO ID	1510301
System Tracking Program ID	System # 5166, T/A # 12807, D/T # 5118
Requirements Source	
Unified Capabilities Requirements	Unified Capabilities Requirements 2013, Errata 1
Remarks	
Test Organization(s)	Joint Interoperability Test Command, Fort Huachuca, Arizona
LEGEND: ID Identification	UCCO Unified Capabilities Connection Office

2. SYSTEM DESCRIPTION. The Session Controller (SC) is a software-based call-processing product that provides voice and video services to Internet Protocol (IP) telephones and media processing devices within a service domain. An SC extends signaling and session (call) control services to allow sessions to be established with users outside a given service domain via an IP-based long-haul network or via gateways to non-IP networks. The SC software and functions may be distributed physically among several high-availability server platforms with redundant call management modules and subscriber tables to provide robustness. Different types of SCs can be deployed, depending upon the service environment. These types are Local, Enterprise, and Master and Subtended. Local SCs (LSCs) are physically located at the Base/Post/Camp/Station (B/P/C/S) where the End Instruments (EIs) they serve are located.

The Enterprise Unified Capabilities (UC) Services Architecture consists of Enterprise Session Controller (ESC) Core Infrastructure products at a centralized “Master Site” location and Edge Infrastructure products at Department of Defense (DoD) Components’ B/P/C/S locations. The ESC Core Infrastructure is composed of centralized ESC components, an ESC-fronting Session Border Controllers (SBC), Enterprise Hosted UC Services and Enterprise Required Ancillary Equipment (RAE). The Edge Infrastructure consists of EIs, Media Gateways (MGs), Enclave-fronting SBCs, local survivable call processing appliances (for Environments 1 and 2) and local RAE. The geographic region that encompasses the centralized ESC location together with all of the served DoD Components B/P/C/S locations is referred to as the Enterprise Services Area (ESA). The ESC provides an integrated management framework that enables the centralized configuration, provisioning, administration, management, and monitoring of all Centralized Enterprise Services Infrastructure components and all Edge Infrastructure components within the ESA. Reliable and redundant systems at all levels of the Enterprise UC Services architecture ensure the high availability needed to meet the requirements of the warfighter and operational user. The ESC provides centralized, integrated voice, video and data session management on behalf of served IP EIs that are located at different enclaves (i.e., B/P/C/S locations) within the served geographic region. A full suite of Enterprise Hosted UC Services are collocated with the ESC. The Hosted UC Services include the following:

- Centralized voice and video session management.
- Centralized voice and video conferencing.
- Unified messaging.
- Integrated E911 Call Management.
- Extensible Messaging and Presence Protocol (XMPP) Instant Messaging (IM)/Chat/Presence federation.
- Service portability.
- Integrated Enterprise Directory Services.

The SUT is certified as an LSC. The SUT is composed of the following components:

- AudioCodes Mediant 3000 (M3000)
- AudioCodes M800
- EXPERiUS Server (x2) (Includes Personal Agent, Session Manager and Media Application Server)
- Management Workstation (site-provided laptop)
- TEO 4101, 4104, and 7810 Assured Services Session Initiation Protocol (AS-SIP) EI (AEI) with Multi-Level Precedence and Preemption (MLPP) (voice only)
- Polycom Video and Voice eXchange (VVX) phones 510 and 610 (ROUTINE only voice and video)
- Polycom VVX Phones 310 and 410 (ROUTINE only voice)
- GENBAND Multimedia client with MLPP (certified for voice only) (site-provided laptop)

EXPERiUS 11.2 Server (x2) - The GENBAND EXPERiUS Application Server is a software-based Session Initiation Protocol (SIP) application server that delivers multimedia applications over any IP broadband to legacy Time Division Multiplexing (TDM) endpoints. The server features Open Programmability Suite with SIP Open Programmable Interface and Representation State Transfer Application Programmable Interfaces. The server hosts the following virtual servers:

- PA (VMWare) (x2) - The PA serves as an end user portal to provide management functions for the EXPERiUS user administered passwords, reservation less conferencing bridges, and unified messaging.

- SESM (VMWare) (x2) - The LSC SESM is the service execution engine that provides the following software functionality: Back-to-Back User Agent, Call processing Language Interpreter, Assured Services Admission Control Budgeting, and address resolution and routing capabilities.

- MAS (VMWare) (x2) - The MAS provides a high availability media services run time base and is packaged with the following applications: Ad hoc Conferencing, Meet Me Conferencing, Tones/Announcements, Music on Hold, Voice Mail. It supports Secure SIP and Secure Real Time Transport secured media. Services provisioning and end user controls settings are through the SIP Core Provisioning Client and PAs. The MAS node management is via the MAS Management Console. Additional MAS servers can be added to meet increased capacity requirements.

AudioCodes 3000 – The M3000 provides Digital Transmission Link Level 1 (T1) Time-Division Multiplexing (TDM) access to the Public Switch Telephone Network (PSTN) Defense Switch Network (DSN).

AudioCodes M800 – The M800 provides analog gateway and PSTN/DSN capability.

TEO IP Phone 7810 – The 7810 is a voice only AEI with IP version 4 (IPv4)\IP version 6 (IPv6) and MLPP support. IPv6 was not tested and is not covered under this certification.

TEO IP Phone 4101 – The 4101 is a voice only AEI with IPv4\IPv6 and MLPP support. IPv6 was not tested and is not covered under this certification.

Polycom VVX Phones 310 and 410 – The Polycom VVX is a ROUTINE only SIP Voice over IP (VoIP) EI.

Polycom VVX Phones 500 and 600 – The Polycom VVX is a ROUTINE only UC media communications end instrument VVoIP client.

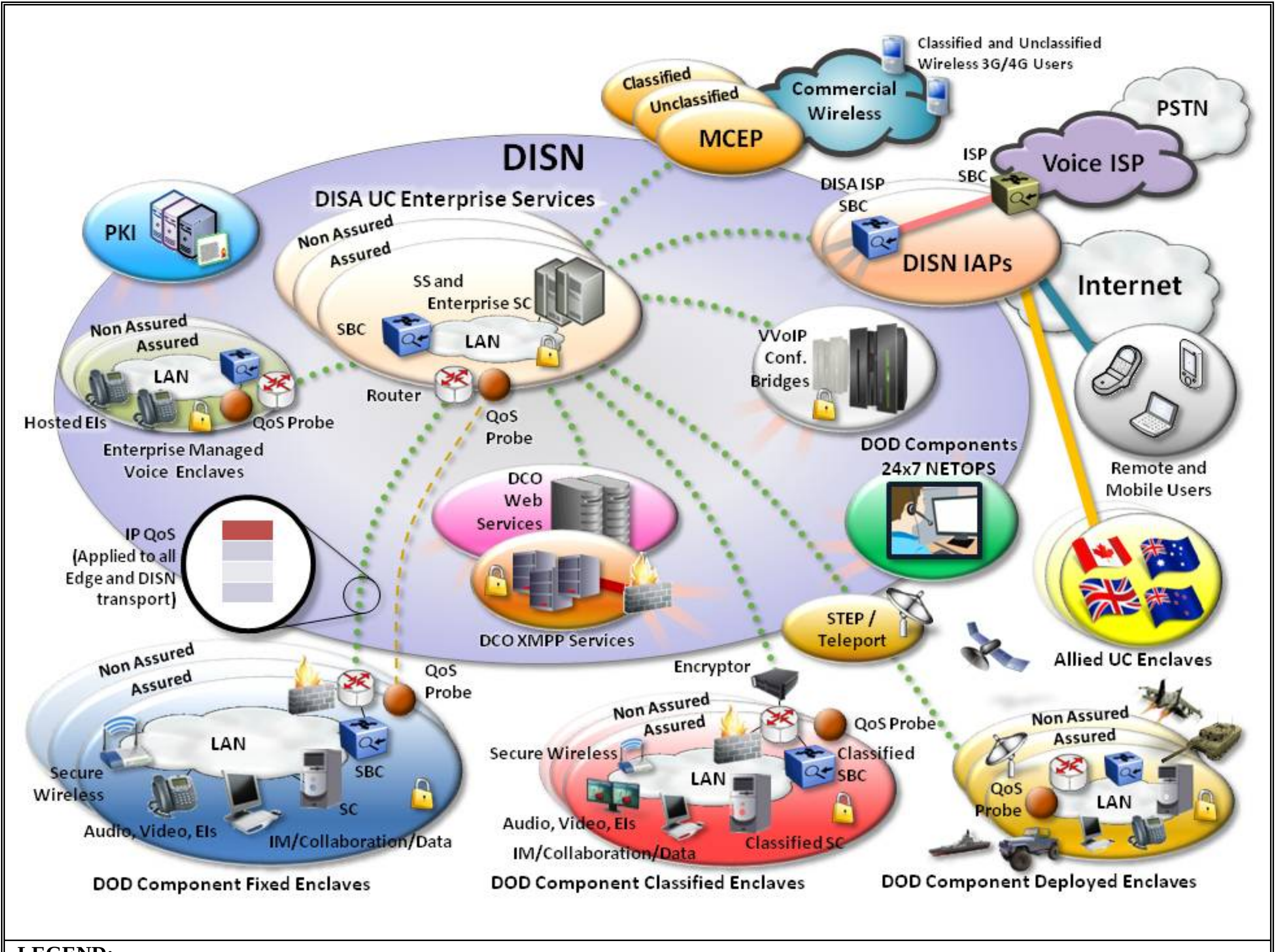
GENBAND Multimedia Client – The GENBAND Multimedia Client is a site-provided, Common Access Card (CAC)-enabled, Secure Technical Implementation Guide (STIG)-compliant Windows 7 workstation that hosts the GENcom soft client. The GENBAND multimedia client supports MLPP for both video and voice; however, only voice is certified for joint use.

Management Workstation – The Management Workstation is a site-provided, CAC-enabled, STIG-compliant Windows 7 workstation. It hosts the system Management Console (MCP), which provides access to the seven virtual applications.

3. OPERATIONAL ARCHITECTURE. The UC architecture is a two-level network hierarchy consisting of Defense Information Systems Network (DISN) backbone switches and Service/Agency installation switches. The Department of Defense (DoD) Chief Information Officer (CIO) and Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The UC architecture, therefore, consists of several categories of switches. Figure 2-1 depicts the notional operational UC architecture in which the SUT may be used and Figure 2-2 the LSC functional model.

4. TEST CONFIGURATION. The test team tested the SUT at JITC, Fort Huachuca, Arizona in a manner and configuration similar to that of a notional operational environment. Testing of the system's required functions and features was conducted using the test configuration depicted in Figure 2-3. Information Assurance testing used the same configuration.

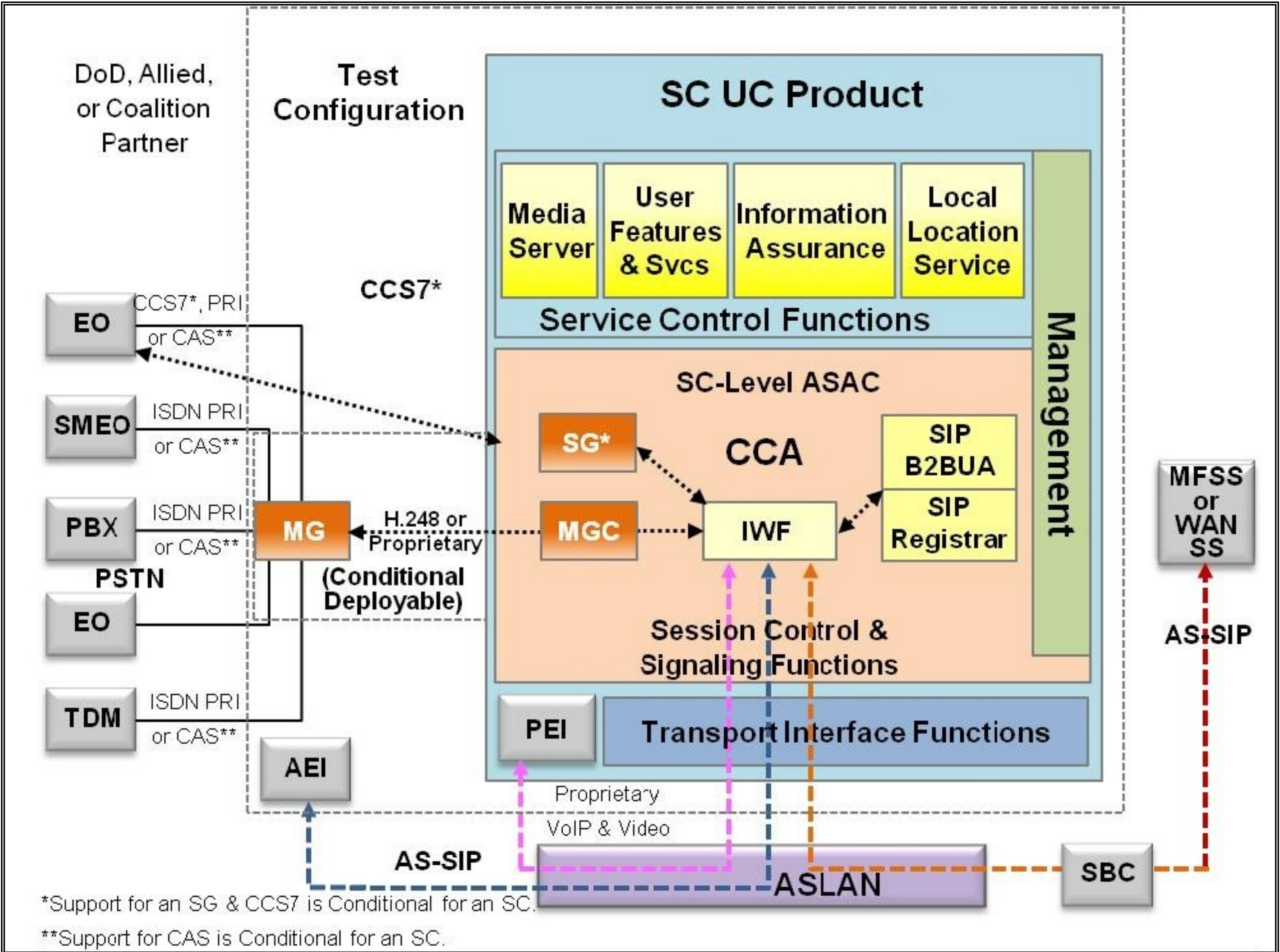
5. METHODOLOGY. Testing was conducted using LSC requirements derived from the Unified Capabilities Requirements (UCR) 2013, Errata 1, Reference (b), and Session Controller test procedures, Reference (c). Any discrepancies noted were documented in Test Discrepancy Reports (TDRs). The vendor submitted Plan of Action and Milestones (POA&M) as required. The TDRs were adjudicated by DISA as minor. Any new discrepancy noted in the operational environment will be evaluated for impact on the existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor POA&M, which will address all new critical TDRs within 120 days of identification.



LEGEND:

DCO	Defense Connection Online	NETOPS	Network Operations
DISA	Defense Information Systems Agency	PKI	Public Key Infrastructure
DISN	Defense Information Systems Network	PSTN	Public Switched Telephone Network
DoD	Department of Defense	QoS	Quality of Service
EI	End Instrument	SBC	Session Border Controller
IAP	Internet Access Point	SC	Session Controller
IM	Instant Messaging	SS	Softswitch
IP	Internet Protocol	STEP	Standardized Tactical Entry Point
ISP	Internet Service Provider	UC	Unified Capabilities
LAN	Local Area Network	VVoIP	Voice and Video over IP
MCEP	Multi Carrier Entry Point	XMPP	Extensible Messaging and Presence Protocol

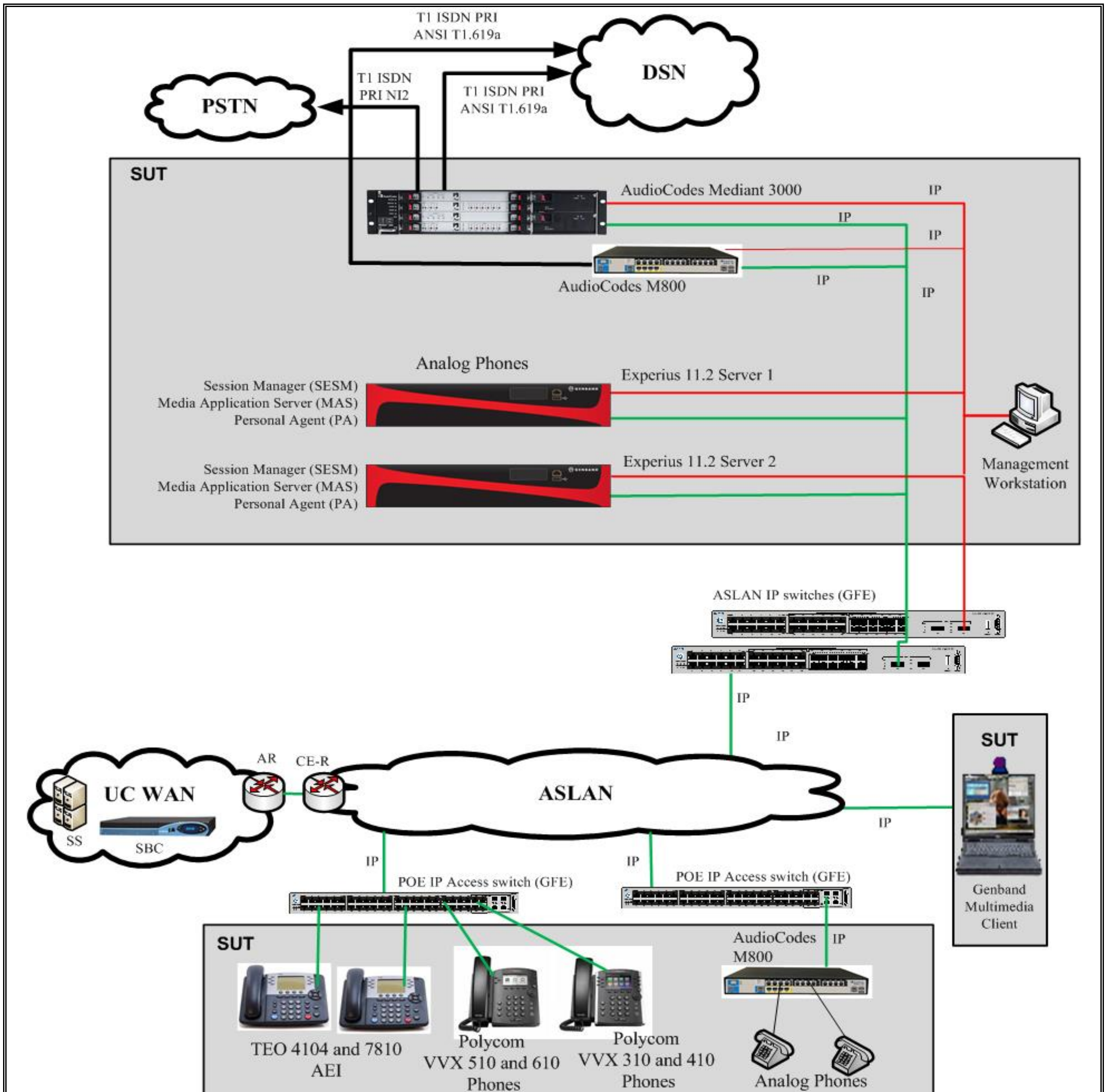
Figure 2-1. Notional UC Network Architecture



LEGEND:

AEI	AS-SIP End Instrument
ASAC	Assured Services Admission Control
ASLAN	Assured Services Local Area Network
AS-SIP	Assured Services Session Initiation Protocol
B2BUA	Back-to-Back User Agent
CAS	Channel Associated Signaling
CCA	Call Connection Agent
CCS7	Common Channel Signaling 7
DoD	Department of Defense
EO	End Office
ISDN	Integrated Services Digital Network
IWF	Interworking Function
LSC	Local Session Controller
MFSS	Multifunction Softswitch
MG	Media Gateway
MGC	Media Gateway Controller
PBX	Private Branch Exchange
PEI	Proprietary End Instrument
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
SBC	Session Border Controller
SC	Session Controller
SG	Signaling Gateway
SIP	Session Initiation Protocol
SMEO	Small End Office
SS	Softswitch
TDM	Time Division Multiplexing
UC	Unified Capabilities
WAN	Wide Area Network

Figure 2-2. LSC Functional Reference Model



LEGEND:

AEI	AS-SIP End Instrument	NI-2	National ISDN 2
ANSI	American National Standards Institute	POE	Power over Ethernet
AR	Aggregation Router	PRI	Primary Rate Interface
ASLAN	Assured Services Local Area Network	PSTN	Public Switched Telephone Network
AS-SIP	Assured Services Session Initiation Protocol	RAE	Required Ancillary Equipment
CE-R	Customer Edge Router	SBC	Session Border Controller
DSN	Defense Switched Network	SS	Softswitch
EI	End Instrument	SUT	System Under Test
ESC	Enterprise Session Controller	T1	Digital Transmission Link Level 1
GFE	Gigabit-Fast Ethernet	UC	Unified Capabilities
IP	Internet Protocol	VVX	Voice and Video eXchange
ISDN	Integrated Services Digital Network	WAN	Wide Area Network
LSC	Local Session Controller		

Figure 2-3. SUT LSC Test Configuration

6. INTEROPERABILITY REQUIREMENTS, RESULTS, AND ANALYSIS. The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for LSCs are established by UCR 2013, Errata 1, sections 2, 4, and 5.

a. The UCR 2013, Errata 1, section 2.2, states that it is expected that all Assured Services products, such as SCs and SSs, will support vendor-proprietary VVoIP features and capabilities. The product's support for these vendor-proprietary VVoIP features and capabilities shall not adversely affect the required operation of the MLPP or ASAC features on that product. In addition, the SC shall meet the requirements in the subparagraphs below.

(1) The UCR 2013, Errata 1, section 2.2.1, includes four types of call forwarding features considered for UC: Call Forwarding Variable (CFV), Call Forwarding Busy Line (CFBL), Call Forwarding - Don't Answer - All Calls (CFDA), and Selective Call Forwarding (SCF). Call forwarding interaction with MLPP and a reminder ring for call forwarding features are optional. The SUT met this requirement with testing and the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.2.2, includes the requirements for MLPP Interactions with Call Forwarding. If a call is forwarded by a CF feature that supports MLPP, the precedence level of the call shall be preserved during the forwarding process. This section includes the following features: Call Forwarding at a Busy Station and Call Forwarding - No Reply at Called Station. The SUT met this requirement with testing.

(3) The UCR 2013, Errata 1, section 2.2.3, includes the requirements for Precedence Call Waiting. The following treatments apply to precedence levels of PRIORITY and above: Busy With Higher Precedence Call, Busy With Equal Precedence Call, No Answer, Line Active With a Lower Precedence Call, and Call Waiting for Single Call Appearance VoIP Phones. The SUT met this requirement with testing and the vendor's LoC.

(4) The UCR 2013, Errata 1, section 2.2.4, includes the requirements for Call Transfer. The two types of call transfers are normal and explicit. A normal call transfer is a transfer of an incoming call to another party. An explicit call transfer happens when both calls are originated by the same subscriber. This section includes the following features: Call Transfer Interaction at Different Precedence Levels and Call Transfer Interaction at Same Precedence Levels. The SUT met this requirement with testing and the vendor's LoC.

(5) The UCR 2013, Errata 1, section 2.2.5, Call Hold, states that the calls on hold shall retain the precedence of the originating call. The SUT met this requirement with testing.

(6) The UCR 2013, Errata 1, section 2.2.6, includes the requirements for Three-Way Calling (TWC). The SUT met this requirement with testing and the vendor's LoC.

(7) The UCR 2013, Errata 1, section 2.2.7, includes the requirements for Hotline Service. The SUT met this requirement with testing and the vendor's LoC.

(8) The UCR 2013, Errata 1, section 2.2.8, includes the requirements for Calling Number Delivery. The SUT met this requirement with testing and the vendor's LoC.

(9) The UCR 2013, Errata 1, section 2.2.9, includes the requirements for Call Pick-Up. The SUT met this requirement with testing and the vendor's LoC.

(10) The UCR 2013, Errata 1, section 2.2.10, includes the requirements for Precedence Call Diversion. The SUT met this requirement with testing.

(11) The UCR 2013, Errata 1, section 2.2.11, includes the requirements for Public Safety Voice Features. This section includes the following features: Basic Emergency Service (911), Tracing of Terminating Calls, Outgoing Call Tracing, Tracing of a Call in Progress, and Tandem Call Trace. The SUT met this requirement with testing and the vendor's LoC. Tandem call trace is optional and was not tested.

b. The UCR 2013, Errata 1, section 2.3, includes the requirements for ASAC in the subparagraphs below.

(1) The UCR 2013, Errata 1, section 2.3.1, includes the requirements for ASAC requirements related to voice. This section includes the following requirements: Voice Session Budget Unit, ASAC States, Session Control Processing with No Directionalization, and SC Session Control Processing with Directionalization. The SUT met this requirement with testing and the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.3.3, includes the requirements for ASAC Requirements for the SC and the SS Related to Video Services. The SUT met this requirement with testing.

c. The UCR 2013, Errata 1, section 2.4, includes the requirements for Signaling Protocols. This section also includes the Signaling Performance Guidelines, which include call setup and call tear-down times as well as guidelines for intra-enclave, inter-enclave and worldwide calls. The SUT met this requirement with testing and the vendor's LoC with the following minor exception. The SUT is certified in the United States, including the Continental United States (CONUS), Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports European Basic Multiplex Rate (E1) interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M. Therefore, the SUT is not certified for joint use outside CONUS in European Telecommunications Standards Institute (ETSI)-compliant countries.

d. The UCR 2013, Errata 1, section 2.5, includes the requirements for Registration and Authentication. Registration and authentication between Network Elements (NEs) shall follow the requirements set forth in Section 4, Information Assurance. Security is tested by the Information Assurance Test Team and the results published in a separate report, Reference (d). This section includes additional Network Time Protocol (NTP) requirements. The SUT met this requirement with testing and the vendor's LoC.

e. The UCR 2013, Errata 1, section 2.6, includes the requirements for SC and SS Failover and Recovery. An SC may be provisioned with direct links to any number of other SCs (i.e. direct tertiary routes) and provisioned with the set of addresses or record served by each SC with which it has a direct tertiary route. Each SC and SBC pairing shall support OPTIONS-based failover by at least one of the following methods under UCR 2013, Errata 1, sections 2.6.1 or 2.6.2.

(1) Section 2.6.1, SC Failover: Alternative A: The SC-Generated OPTIONS Method. This section covers SC-Generated OPTIONS, SC OPTIONS-based Failover, SC-based Failback and Failure Handling for Outbound Dialog-initiating INVITE. The SUT did not meet this requirement.

(2) Section 2.6.2, SC Failover: Alternative B: The SBC-Generated OPTIONS Method. This section covers SBC-Generated OPTIONS, SBC-based Failover, SBC-based Failback and Failure Handling for Outbound Dialog-initiating INVITE. The SUT met this requirement with testing and the vendor's LoC. The SUT however, does not send Option Pings but does respond to Option Pings, however this discrepancy did not impact Alternative B failover. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

f. The UCR 2013, Errata 1, section 2.7, includes the requirements for Product Interfaces. This includes requirements for internal and external physical interfaces, interfaces to other networks, and DISA VVoIP EMS Interface.

(1) The UCR 2013, Errata 1, section 2.7.1, states that internal interfaces are functions that operate internal to a System Under Test (SUT) or UC-approved product (e.g., SC, SS). The interfaces between SC/SS functions within an SC and Signaling Gateway (SG) are considered internal to the SC regardless of the physical packaging. These interfaces are vendor-proprietary and unique, especially the protocol used over the interface. Whenever the physical interfaces use Institute of Electrical and Electronics Engineers, Inc. (IEEE) 802.3 Ethernet standards, they shall support auto-negotiation even when the IEEE 802.3 standard has it as optional. This applies to 10/100/1000-T Ethernet standards; i.e., IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995; and IEEE, Gigabit Ethernet Standard 802.3ab, 1999. The vendor met this requirement with the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.7.2, states External Physical Interfaces Between Network Components are functions that cross the demarcation point between SUT and other external network components. Whenever the physical interfaces use IEEE 802.3 Ethernet standards, they shall support auto-negotiation even when the IEEE 802.3 standard has it as optional. This applies to 10/100/1000-T Ethernet standards; i.e., IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995; and IEEE, Gigabit Ethernet Standard 802.3ab, 1999. This requirement does not preclude the use of other types of physical interfaces. The vendor met this requirement with the vendor's LoC.

(a) The SC (and its appliances), SS, and SBC shall support 10/100/1000-T Mbps Ethernet physical interfaces to ASLAN switches and routers. The vendor met this requirement with the vendor's LoC.

(b) In addition, PEI and AEI shall support 10/100-T Mbps Ethernet physical interfaces to ASLAN switches and routers. The vendor met this requirement with the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.7.3, states that interfaces to other networks, are interfaces where traffic flows from one network (e.g., UC) to another network (e.g., PSTN). The SUT met this requirement with testing and the vendor's LoC.

(a) The Deployable interface requirements are specified in Appendix A, Unique Deployed (Tactical), and Section 6, Network Infrastructure End-to-End Performance. These requirements are unique optional requirements for a deployed SC and were not tested.

(b) The Assured Services subsystem shall interface the Teleport sites on both a TDM basis and an IP basis. An ANSI T1.619a MG with PRI signaling will be used for T1 trunks to the Teleport sites. If the Teleport site contains an SC, then the interface will be via the DISN WAN for both the media and signaling, with the signaling being AS-SIP (AS-SIP 2013) between the Teleport SC and the UC SS. The SUT met this requirement with testing and the vendor's LoC.

(c) The Assured Services subsystem shall interface with the PSTN and host-nation PTTs via the MG interfaces as specified in Section 2.16, Media Gateway. The SUT met this requirement with testing and the vendor's LoC.

(d) Voice and video interfaces with allied and coalition networks have not yet been defined. Therefore, the interface will remain TDM as specified in UCR 2013, Figure 4.4.2-1, DSN Design and Components.

(4) The UCR 2013, Errata 1, section 2.7.4, states that SCs shall support a 10/100-Mbps Ethernet physical interface to the DISA VVoIP EMS. The interface will work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. The SUT met this requirement with testing and the vendor's LoC.

g. The UCR 2013, Errata 1, section 2.8, includes the requirements for the Product Physical, Quality and Environmental Factors.

(1) The UCR 2013, Errata 1, section 2.8.2.1, includes the Product Availability requirements. This includes requirements for the following three availability options: High Availability, Medium Availability, and Medium Availability for non-Command and Control (C2) users. The SUT met the high-availability requirement with testing and the vendor's LoC.

(a) High Availability. The Assured Services appliance shall have a hardware or software availability of 0.99999 (a nonavailability state of no more than 5 minutes per year). The vendor shall provide an availability model for the appliance showing all calculations and showing how the overall availability will be met. The subsystem(s) shall have no single point of failure that can cause an outage of more than 96 voice and/or video subscribers. To meet the availability requirements, all subsystem(s) platforms that offer service to more than 96 voice and/or video subscribers shall have a modular chassis that provides, at a minimum, the following: Dual Power Supplies, Dual Processors/Swappable Sparing (Control Supervisors), Termination Sparing, Redundancy Protocol, No Single Failure Point, Switch Fabric or Backplane Redundancy for Active Backplanes, Software Upgrades and Patches, Backup Power UPS Requirements, No Loss of Active Sessions. In addition, when an appliance component fails and the backup component takes over, each media stream for each active call shall remain active during the failover, until either 1) timer expirations or lack of state information cause that call to terminate or 2) the EI subscribers on that call, naturally terminate the call.

(b) Medium Availability. The Medium Availability SC shall have a hardware or software availability of 0.9999 (a non-availability state of no more than 53 minutes per year). The vendor shall provide an availability model for the appliance showing all calculations and showing how the overall availability will be met. In support of the availability requirements, all subsystem(s) platforms that offer service to more than 96 voice and/or video subscribers shall have a modular design and chassis that provides, at a minimum, the following: Dual Power Supplies, Software Upgrades and Patches, No Loss of Active Sessions. In addition, when an Appliance component fails and the backup component takes over, each media stream for each active call shall remain active during the failover, until either 1) timer expirations or lack of state information cause that call to terminate or 2) the EI subscribers on that call naturally terminate the call.

(c) Medium Availability not for Special C2 users. The Medium Availability SC shall have a modular design and chassis that provides, at a minimum, the following: Dual Processors/Swappable Sparing (Control Supervisors), Termination Sparing, Redundancy Protocol, No Single Failure Point, Switch Fabric or Backplane Redundancy for Active Backplanes, and Backup Power UPS Requirements.

(2) The UCR 2013, Errata 1, section 2.8.2.2, includes the Maximum Downtime requirements. This includes requirements for the following products: High Availability and Medium Availability. The SUT met this requirement with the vendor's LoC.

(a) High Availability. The performance parameters associated with the ASLAN, SS, and High Availability SC, when combined, shall meet the following maximum downtime requirements: IP (10/100/1000 Ethernet) network links - 35 minutes per year and IP subscriber - 12 minutes per year.

(b) Medium Availability. The performance parameters associated with the ASLAN, SS, and Medium Availability SC, when combined, shall meet the following maximum downtime requirements: IP (10/100/1000 Ethernet) network links - 82 minutes per year and IP subscriber – 60 minutes per year.

(3) The UCR 2013, Errata 1, section 2.8.4, states that, for the VoIP devices, the voice quality shall have a Mean Opinion Score (MOS) of 4.0 (R-Factor equals 80) or better, as measured IAW the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period. This applies only to devices that generate media and have a Network Interface Card (NIC). The SUT met this requirement with testing and the vendor's LoC.

8. The UCR 2013, Errata 1, section 2.9, includes the requirements for End Instruments (EIs) in the subparagraphs below. The SUT met these requirements with testing and the vendor's LoC with the following minor exceptions. The Polycom 310, 410, 510, and 610 EIs do not divert precedence above ROUTINE calls to an attendant or Alternate DN. DISA has adjudicated this discrepancy as minor. This requirement was changed in the UCR 2013, Change 1 to allow precedence calls above ROUTINE to be answered by an ROEI if resources are available.

h. During testing, the SUT soft client was unable to establish two-way video with other video systems. DISA has accepted the vendor's POA&M and adjudicated this as critical for video on the soft client. The soft client is certified for audio only.

(1) The UCR 2013, Errata 1, section 2.9.1, includes the requirements for IP Voice End Instruments. This section includes the basic requirements, tones and announcements, audio codecs for voice instruments, VoIP telephone audio performance, VoIP sampling standard, softphones, and DSCP packet marking. The SUT met this requirement with testing and the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.9.2, includes the requirements for analog and ISDN BRI Telephone Support. The SUT met this requirement with testing and the vendor's LoC. BRI end instruments are conditional and not supported by the SUT.

(3) The UCR 2013, Errata 1, section 2.9.3, includes the requirements for Video End Instruments. Video EIs are considered associated with the SC and must have designated in conjunction with the SC design. This section includes the basic requirements; display messages, tones, and announcements; video codecs including associated audio codecs; and H.323 video teleconferencing. The SUT met this requirement with testing and the vendor's LoC.

(4) The UCR 2013, Errata 1, section 2.9.4, states that the PEI and AEI shall each be capable of authenticating itself to its associated SC and vice versa IAW Section 4, Information Assurance. The SUT met this requirement with testing and the vendor's LoC. In addition, security is tested by the Information Assurance Test Team and the results published in a separate report, Reference (d).

(5) The UCR 2013, Errata 1, section 2.9.5, states that the EI interface to the ASLAN shall be IAW Ethernet (IEEE 802.3) Local Area Network (LAN) technology. The 10-Mbps and 100-Mbps Fast Ethernet (IEEE 802.3u) shall be supported. The SUT met this requirement with the vendor's LoC.

(6) The UCR 2013, Errata 1, section 2.9.6, includes the requirements for Operational Framework for AEIs. This section contains SC and AS-SIP EI requirements to support a generic, multivendor-interoperable interface between a VVoIP SC and an AS-SIP VVoIP EI, which can be a voice EI, a secure voice EI, or a video EI. This generic, multivendor-interoperable interface uses AS-SIP protocol instead of the various vendor-proprietary SC-to-EI protocols. This section includes the following requirements: Requirements for Supporting AS-SIP EIs, Requirements for AS-SIP voice EIs, Requirements for AS-SIP Secure voice EIs, Requirements for AS-SIP video EIs, and AS-SIP video EI features. The SUT met this requirement with testing and the vendor's LoC with the following minor exception. The SUT Polycom VVX 510 and 610 video phones were interoperable with other tested video endpoints listed on the UC APL except for the Cisco 99xx video EIs. When the Cisco 99xx series video phones originate a video session with the SUT VVX 510 and 610 video phones it results in two-way audio and one-way video (VVX has no video). DISA has accepted the vendor's POA&M and adjudicated this discrepancy as minor.

(7) The UCR 2013, Errata 1, section 2.9.7, includes the requirements for Multiple Call Appearance Requirements for AS-SIP EIs. This section includes the following requirements: Multiple Call Appearance Scenarios, Multiple Call Appearances - Specific Requirements, and Multiple Call Appearances - Interactions With Precedence Calls. The SUT met this requirement with testing and the vendor's LoC.

(8) The UCR 2013, Errata 1, section 2.9.8 includes the requirements for PEIs, AEIs, TAs, and IADs using the ITU-T Recommendation V.150.1 protocol. The SUT met this requirement with testing and the vendor's LoC.

(9) The UCR 2013, Errata 1, section 2.9.9 includes the requirements for UC Products with Non-Assured Service Features. The vendor met this requirement with testing and the vendor's LoC.

9. The UCR 2013, Errata 1, section 2.9.10 includes the requirements for ROUTINE-Only EIs (ROEIs). SC support for ROEIs is optional; however, if they are supported the requirements in this section apply. The SUT met this requirement with testing with the following minor exception. The Polycom 310, 410, 510, and 610 EIs do not divert precedence above ROUTINE calls to an attendant or Alternate DN. DISA has adjudicated this discrepancy as minor. This requirement was changed in the UCR 2013, Change 1 to allow precedence calls above ROUTINE to be answered by an ROEI if resources are available.

i. The UCR 2013, Errata 1, section 2.10 includes the requirements for Session Controllers.

(1) The UCR 2013, Errata 1, section 2.10.1, states the SC shall meet all the requirements for Precedence-Based Assured Services (PBAS) and ASAC, as appropriate for VoIP and Video over IP services, as specified in Section 2.25.1, Multilevel Precedence and Preemption and Section 2.3, ASAC. The SUT met this requirement with testing.

(2) The UCR 2013, Errata 1, section 2.10.2 states the SC shall support AS-SIP over IP for signaling to SSs. The SC shall optionally support AS-SIP over IP signaling to AEIs. The SC

shall optionally support proprietary VVoIP signaling to interface with PEI. The SUT met this requirement with testing and the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.10.3, states that the SC shall support a Session Controller Location Service (SCLS) functionality that provides information on call routing and called address translation. The SUT met this requirement with testing.

(4) The UCR 2013, Errata 1, section 2.10.4, states that the SC shall support the applicable Fault, Configuration, Accounting, Performance, and Security (FCAPS) Management and Audit Log requirements specified in Section 2.19, Management of Network Appliances. The SUT met this requirement with the vendor's LoC.

(5) The UCR 2013, Errata 1, section 2.10.5, states that the SC shall provide an interface to the DISA VVoIP Element Management System (EMS). The interface consists of a 10/100-Mbps Ethernet connection as specified in Section 2.7.4, DISA VVoIP EMS Interface. The SUT met this requirement with the vendor's LoC.

(6) The UCR 2013, Errata 1, section 2.10.6, states that the SC shall provide Transport Interface functions to interface with the ASLAN and its IP packet transport network. The SUT met this requirement with testing and the vendor's LoC.

(7) The UCR 2013, Errata 1, section 2.10.7, includes the requirements for Custom Line-Side Features Interference. If custom line-side features are implemented, they shall not interfere with the Assured Services requirements. The SUT was not tested with any Custom Line-Side Features and they are not covered under this certification.

(8) The UCR 2013, Errata 1, section 2.10.8, states that during the call establishment process, the product shall be capable of preventing or detecting and stopping hairpin routing loops over American National Standards Institute (ANSI) T1.619a and commercial PRI trunk groups between a legacy switch and an SC. The Loop Avoidance mechanism shall not block call requests that are legitimately redirected or forwarded between the two interconnected products. In the event that a routing loop is detected, the SC shall clear the call in the backwards direction, either sending a 404 (Not Found) response to a SIP originator, or an ISDN DISCONNECT message (from the MG) to a TDM originator. The SC shall provide a VCA to the caller in each case. The SUT met this requirement with testing.

(9) The UCR 2013, Errata 1, section 2.10.9, states that the requirements in this section are unique to the distributed application for the Session Controller, the Local Session Controller (LSC). An LSC provides management for the EIs within a single B/P/C/S or enclave and is deployed at the same location as the EIs it serves. The SUT met this requirement with testing.

(10) The UCR 2013, Errata 1, section 2.10.10, states that in the event that total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions: Completion of local calls (i.e., calls between EIs the LSC serves); Routing of calls to and from the PSTN, using a local MG (PRI or CAS as required by the local interface), that originate or

terminate on EIs the LSC serves; User look-up of local directory information. The SUT met this requirement with testing.

j. The UCR 2013, Errata 1, section 2.11, includes requirements for the AS-SIP Gateways.

(1) The UCR 2013, Errata 1, section 2.11.1, includes requirements for the AS-SIP TDM Gateway Signaling. The SUT met this requirement with testing as a component of the SUT.

(2) The UCR 2013, Errata 1, section 2.11.2, includes requirements for the AS-SIP IP Gateway. The SUT does not support this optional requirement.

(3) The UCR 2013, Errata 1, section 2.11.3, includes requirements for the AS-SIP – H.323 Gateway. The SUT does not support this optional requirement.

k. The UCR 2013, Errata 1, section 2.14, includes requirements for the Call Connection Agent (CCA) function in the SC and SS. Both of these appliances have a DISN-defined design that includes Session Control and Signaling functions. These functions include both a Signaling Protocol IWF and an MGC function. The CCA described in the following requirements is part of the SCS functions, and includes both the IWF and the MGC.

(1) The UCR 2013, Errata 1, section 2.14.1, includes requirements for the CCA in an SS or SC to be able to support multiple MGs on a single ASLAN, multiple ASLANs, or multiple physical locations. The SUT met this requirement with the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.14.2, includes the Functional requirements of the CCA to include the IWF and MGC Component. The SUT met this requirement with the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.14.4, includes requirements for CCA-IWF Signaling Protocol Support. This section includes the requirements for the CCA Signaling Protocol IWF to support the various VoIP and TDM signaling protocols used in the SC and SS. The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that are used by the EIs, MGs, and SBCs, and interwork all these various signaling protocols with one another. The SUT met this requirement with testing and the vendor's LoC.

(a) The UCR 2013, Errata 1, section 2.14.4.1, includes the requirements for CCA-IWF support for AS-SIP. The SUT met this requirement with the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.14.4.2, includes the requirements for CCA-IWF Support for PRI via MG. The SUT met this requirement with testing and the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.14.4.3, includes the optional requirements for CCA-IWF Support for CAS Trunks via MG. Although the SUT supports this conditional requirement it was not tested and is not covered under this certification.

(d) The UCR 2013, Errata 1, section 2.14.4.4, includes the requirements for CCA-IWF Support for PEI and AEI Signaling Protocols. The SUT met this requirement with the vendor's LoC.

(e) The UCR 2013, Errata 1, section 2.14.4.5, includes the requirements for CCA-IWF Support for VoIP and TDM Protocol Interworking. The SUT met this requirement with the vendor's LoC for T1 PRI. The SUT is certified in the United States, including the Continental United States (CONUS), Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports European Basic Multiplex Rate (E1) interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated as minor with vendor POA&M with the following condition of fielding: The SUT is not certified for joint use outside CONUS in European Telecommunications Standards Institute (ETSI)-compliant countries. Although the SUT supports CAS, it was not tested and is not covered under this certification.

l. The UCR 2013, Errata 1, section 2.15, includes the requirements for CCA Interaction with Network Appliances and Functions. The SUT met this requirement with testing and the vendor's LoC.

(1) The UCR 2013, Errata 1, section 2.15.1, includes the requirements for CCA interaction with transport interface functions. The transport interface functions in an appliance provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met this requirement with the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.15.2, includes the requirements for CCA interactions with the SBC requirements. The SUT met this requirement with the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.15.3, includes the requirements for CCA support for Admission Control requirements. The CCA interacts with the ASAC component of the SC and SS to perform specific functions related to ASAC, such as counting internal, outgoing, and incoming calls; managing separate call budgets for VoIP and Video over IP calls; and providing preemption. The SUT met this requirement with the vendor's LoC.

(4) The UCR 2013, Errata 1, section 2.15.4, includes the requirements for CCA support for User Features and Services. The User Features and Services (UFS) Server is responsible for providing features and services to VoIP and Video PEIs/AEIs on an SC or SS, where the CCA alone cannot provide the feature or service. This requirement states that the CCA within a network appliance shall support the operation of the following features and capabilities: The CCA shall generate a redirecting number each time it forwards a VoIP or Video session request as part of a CF feature. The CCA supports the ability to direct VoIP and Video sessions and session requests to the UFS Server, so that the UFS Server can apply an Appliance VoIP or Video feature, when use of that feature is required by the calling party, the called party, or the appliance itself. The SUT met this requirement with the vendor's LoC.

(5) The UCR 2013, Errata 1, section 2.15.5, includes the requirements for CCA Support for Information Assurance. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, and SBCs that use the appliance are all properly authenticated

and authorized by the appliance. The Information Assurance function ensures that voice and video signaling streams that traverse the appliance and its ASLAN are properly encrypted SIP/TLS. The SUT met this requirement with the vendor's LoC. In addition, security is tested by a separate Information Assurance Test Team and the results published in a separate report, Reference (d).

(6) The UCR 2013, Errata 1, section 2.15.8, includes the requirements for CCA interactions with EIs. The CCA in the SS or SC needs to interact with VoIP PEIs and AEIs served by that SS or SC. The VoIP interface between the PEI and the SS or SC is left up to the network appliance supplier. The VoIP interface between the AEI and the SS or SC is AS-SIP. The SUT met this requirement with the vendor's LoC.

(7) The UCR 2013, Errata 1, section 2.15.9, includes the requirements for CCA support for Assured Services voice and video. The Appliance CCA shall support both assured voice and video services. The CCA shall support both assured voice and assured video sessions, and shall support these sessions from both, VoIP EIs and Video EIs, as described in Section 2.15.8, CCA Interactions with End Instrument(s). The SUT met this requirement with the vendor's LoC.

(8) The UCR 2013, Errata 1, section 2.15.10, includes the requirements for CCA interactions with service control functions. This requirement states that the CCA shall support the ability to remove VoIP and video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. The SUT met this requirement with the vendor's LoC.

m. The UCR 2013, Errata 1, section 2.16, includes requirements for the MG function in the SC and SS network appliances. These appliances have defined designs that include a Media Gateway Controller (MGC) function and one or more MGs. The SUT met these requirements with testing and the vendor's LoC with the following minor exception. The SUT is certified in the United States, including the CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.

(1) The UCR 2013, Errata 1, section 2.16.1, includes the requirements for MG call denial treatments to support CAC requirements. When the CCA determines that a VoIP session request should be blocked because an Appliance CAC restriction applies, the CCA will deny the session request and apply a Call Denial treatment to the calling party on that request. The SUT met this requirement with testing and the vendor's LoC.

(a) The UCR 2013, Errata 1, section 2.16.1.1, includes the requirements for MG call preemption treatments to Support ASAC. The SUT met this requirement with testing and the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.16.1.2, includes the requirements for MG and Information Assurance functions. Information Assurance is tested by a separate Information

Assurance Test Team and the results published in a separate report, Reference (d). The SUT met this requirement with the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.16.1.3, states that the media server is responsible for playing tones and announcements to calling and called parties on VoIP calls, and for playing audio/video clips (similar to tones and announcements) to calling and called parties on video calls. The MG is responsible for routing individual VoIP, FoIP, and MoIP media streams to the media server when instructed to do so by the CCA/MGC. When instructed to do so by the MGC, the MG shall direct TDM calls and call requests to the media server, so that the media server can do the following: Play tones and announcements to TDM parties on TDM calls and call requests (e.g., busy tone or announcement; call preemption tone or announcement). Provide "play announcement and collect digits" functionality when required by an Appliance VoIP feature. Provide full IVR-like functionality when required by an Appliance VoIP feature. The SUT met this requirement with testing and the vendor's LoC.

(d) The UCR 2013, Errata 1, section 2.16.1.4, includes the requirements for interactions with IP Transport interface functions. The SUT met this requirement with testing and the vendor's LoC.

(e) The UCR 2013, Errata 1, section 2.16.1.5, includes the requirements for MG–SBC interaction. The SUT met this requirement with testing and the vendor's LoC.

(f) The UCR 2013, Errata 1, section 2.16.1.6, states the Management function in the SBC, SC, and SS supports functions for SBC/SC/SS FCAPS management and audit logs. The MG shall interact with the Appliance Management function by doing the following: Making changes to its configuration and to its trunks' configuration in response to commands from the Management function that request these changes. Returning information to the Management function on its FCAPS in response to commands from the Management function that request this information. Sending information to the Management function on a periodic basis (e.g., on a set schedule), keeping the Management function up-to-date on MG activity. The SUT met this requirement with the vendor's LoC.

(g) The UCR 2013, Errata 1, section 2.16.1.8, states the MG shall support the exchange of VoIP media streams with the following voice PEIs and AEIs both on the local appliance and on remote network appliances: Vendor proprietary Voice PEIs; Voice SIP EIs, when the appliance supplier supports these EIs; Voice H.323 EIs, when the appliance supplier supports these EIs; Voice AS-SIP EIs. The SUT met this requirement with testing and the vendor's LoC.

(h) The UCR 2013, Errata 1, section 2.16.1.9, states the MG shall support the operation of the following features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches: Call Hold, Music on Hold, Call Waiting, Precedence Call Waiting, Call Forwarding Variable, Call Forwarding Busy Line, Call Forwarding No Answer, Call Transfer, Three-way calling, Hotline Service, Calling Number Delivery, and Call Pickup. The SUT met this requirement with testing and the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.16.2, includes requirements for MG interfaces to TDM NEs in DoD networks. Each appliance MG shall support TDM trunk groups that can interconnect with the following NEs in DoD networks, in the United States and worldwide: PBXs, SMEOs, EOs, and MFSs. Each appliance MG shall support TDM trunk groups that can interconnect with DISN and DoD NEs in the United States and worldwide using ISDN PRI ANSI T1.619a. The appliance MG may optionally support CAS. The SUT met this requirement for ISDN PRI, with testing and the vendor's LoC. Although the SUT supports CAS, it was not tested and is not covered under this certification.

n. The UCR 2013, Errata 1, section 2.16.3, includes requirements for MG interfaces to TDM NEs in Allied and Coalition Partner Networks. The appliance suppliers should support TDM trunk groups on their MG product that can interconnect with NEs in U.S. Allied and Coalition partner networks worldwide. The MG shall support foreign country ISDN PRI trunk groups where the MG handles both the media channels and the signaling channel as follows: For interconnection with an allied or coalition partner network, using foreign ISDN PRI from the network of the allied or coalition partner. Support for MLPP using ISDN PRI, per ITU-T Recommendation Q.955.3, is required on SC trunk groups when these trunk groups are used to connect to an allied or coalition partner from a U.S. OCONUS ETSI-compliant country. The SUT met this requirement with testing.   Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.

(3) The UCR 2013, Errata 1, section 2.16.4, includes MG Interfaces to TDM NEs in the PSTN in the United States. Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States, including CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Each appliance MG shall support TDM trunk groups that can interconnect with the U.S. PSTN, using the following types of trunk groups: U.S. National ISDN PRI, where the MG handles both the media channels and the signaling channel is required for U.S. PSTN nationwide; support for FAS is required; support for MLPP is not required; support for NFAS is optional. CAS is optional for U.S. PSTN NEs nationwide. The SUT met this requirement with testing and the vendor's LoC.

o. The UCR 2013, Errata 1, section 2.16.5, includes requirements for MG interfaces to TDM NEs in OCONUS PSTN Networks. The appliance supplier (i.e., SC or SS supplier) should support TDM trunk groups on its MG product that can interconnect with NEs in foreign country PSTN networks (OCONUS) worldwide. The MG shall support foreign country ISDN PRI, where the MG handles both the media channels and the signaling channel: For interconnection with a foreign country PSTN, using foreign country ISDN PRI, from the country where the DoD user's B/P/C/S is located. Support for ETSI PRI is required on SC trunk groups when the SC is used in OCONUS Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.

(4) The UCR 2013, Errata 1, section 2.16.6, includes the requirements for MG support for ISDN PRI trunks.   The SUT met this requirement with testing and the vendor's LoC.

(5) The UCR 2013, Errata 1, section 2.16.7, includes the optional requirements for MG support for CAS trunks. The SUT does not support this optional requirement.

(6) The UCR 2013, Errata 1, section 2.16.8, includes the requirements for MG requirements: VoIP Interfaces Internal to an Appliance. The requirements in this section assume that a supplier-specific Gateway Control Protocol is used on the MGC-MG interface. The SUT met this requirement with testing and vendor's LoC.

(7) The UCR 2013, Errata 1, section 2.16.9, includes the requirements for Echo Cancellation. The requirements in this section are based on the commercial VoIP network Echo Cancellation requirements in Telcordia Technologies GR-3055-CORE. The SUT met this requirement with testing and vendor's LoC.

(8) The UCR 2013, Errata 1, section 2.16.10, includes the requirements for MG requirements for clock timing. The SUT met this requirement with testing and the vendor's LoC.

(9) The UCR 2013, Errata 1, section 2.16.11, includes the requirements for MGC-MC CCA Functions. The MGC within the CCA shall be responsible for controlling all the MGs within the SC or SS. The MGC within the CCA shall be responsible for controlling all the trunks (i.e., PRI or CAS) within each MG within the SC or SS. The MGC within the CCA shall be responsible for controlling all media streams on each trunk within each MG. The SUT met this requirement with testing and the vendor's LoC.

(10) The UCR 2013, Errata 1, section 2.16.12, states that the MG uses ITU-T Recommendation V.150.1, then the following applies: ITU-T Recommendation V.150.1 provides for three states: audio, VBD, and modem relay. After call setup, inband signaling may be used to transition from one state to another. In addition, V.150.1 provides for the transition to FoIP using Fax Relay per ITU-T Recommendation T.38. When the MG uses V.150.1 inband signaling to transition between audio, FoIP, modem relay, or VBD states or modes, the MG shall continue to use the established session's protocol (e.g., decimal 17 for UDP) and port numbers so that the transition is transparent to the SBC. During testing, the AudioCodes MG3K and M800 could not dynamically invoke VBD G.711 or V.150.1. The SUT supports V.150.1 however when G.711 Voice Codec is negotiated the SUT fails to support bi-directional secure calls in pass-through mode. The SUT DISA has accepted the vendor's POA&M and adjudicated this as minor with the Condition of Fielding that the SUT cannot operate with V.150 until this discrepancy is corrected.

(11) The UCR 2013, Errata 1, section 2.16.13, includes the conditional requirements for Remote Media Gateway. If the MG is geographically separated from the MGC that controls it, then the five specific conditional requirements in this section that address the SBC, the MG control protocol, the DSCP for the control packets, and the security aspects for that arrangement apply. The SUT does not comply with this conditional requirement per the vendor's LoC.

p. The UCR 2013, Errata 1, section 2.18, includes the requirements for the Worldwide Numbering and Dialing Plan (WWNDP).

(1) The UCR 2013, Errata 1, section 2.18.1, includes the requirements for DSN Worldwide Numbering and Dialing Plan. The SUT met this requirement with testing and the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.18.1.1, includes the requirements for CCA and SSLS support for dual assignment of DSN and E.164 numbers to SS EIs. The SUT met this requirement with testing and the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.18.1.2, includes the requirements for CCA differentiation between DSN numbers and E.164 numbers. The SUT met this requirement with testing and the vendor's LoC.

(4) The UCR 2013, Errata 1, section 2.18.1.3, includes the optional requirements for CCA use of SIP "phone-context" to differentiate between DSN and E.164 numbers. The SUT met this requirement with testing and the vendor's LoC.

(5) The UCR 2013, Errata 1, section 2.18.1.4, includes the optional requirements for use of SIP URI domain name with DSN numbers and E.164 numbers. The SUT met this requirement with testing and the vendor's LoC.

(6) The UCR 2013, Errata 1, section 2.18.1.5, includes the requirements for Domain Directory. The SUT met this requirement with the vendor's LoC with the following minor exceptions. Per the vendor's LoC, TEO AEIs do not support integration with directory. DISA has adjudicated this discrepancy as minor. During testing, the SUT did not support Domain Directory per the requirement. DISA has accepted the vendor's POA&Ms and adjudicated these discrepancies as minor.

q. The UCR 2013, Errata 1, section 2.19, includes the requirements for the management of network appliances. The SUT met this requirement with the vendor's LoC.

(1) The UCR 2013, Errata 1, section 2.19.1, includes the general management requirements for the management of network appliances. This requirement was met with the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.19.2, includes the requirements for FCAPS Management in the following subparagraphs. General requirements for the five management functional areas are defined.

(a) The UCR 2013, Errata 1, section 2.19.2.1, includes the requirements for Fault Management. The SUT met this requirement with the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.19.2.2, includes the requirements for Configuration Management. The vendor met this requirement with the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.19.2.3, includes the requirements for Accounting Management. The SUT met this requirement with the vendor's LoC.

(d) The UCR 2013, Errata 1, section 2.19.2.4, includes the requirements for Performance Management. The SUT met this requirement with testing and the vendor's LoC.

(e) The UCR 2013, Errata 1, section 2.19.2.5, states that all network management interactions shall meet the access control, confidentiality, integrity, availability, and non-repudiation requirements in Section 4, Information Assurance. Information Assurance is tested by separate IA test teams and the results published in a separate report, Reference (d).

r. The UCR 2013, Errata 1, section 2.20, includes the conditional requirements for V.150.1 modem relay secure phone support. The SUT met this requirement with testing and vendor's LoC with the following exception: During testing, the AudioCodes MG3K and M800 could not dynamically invoke VBD G.711 or V.150.1. The SUT supports V.150.1 however when G.711 Voice Codec is negotiated the SUT fails to support bi-directional secure calls in pass-through mode. The SUT DISA has accepted the vendor's POA&M and adjudicated this as minor with the Condition of Fielding that the SUT cannot operate with V.150 until this discrepancy is corrected.

s. The UCR 2013, Errata 1, section 2.21, includes the conditional requirements for supporting AS-SIP-based Ethernet interfaces for voicemail systems. The SUT met this requirement with the vendor's LoC.

t. The UCR 2013, Errata 1, section 2.22, includes the optional requirements for Local Attendant Console Features. The SUT does not support this optional requirement.

u. The UCR 2013, Errata 1, section 2.23, includes the optional requirements for Master Session Controllers (MSCs) and Subtended Session Controllers (SSCs). This optional configuration was not tested and is not covered under this certification.

v. The UCR 2013, Errata 1, section 2.24, includes the optional requirements for MSC, SSC, and dynamic ASAC requirements in support of bandwidth-constrained links. This optional configuration was not tested and is not covered under this certification.

w. The UCR 2013, Errata 1, section 2.25, includes the requirements for Multilevel Precedence and Preemption (MLPP).

(1) The UCR 2013, Errata 1, section 2.25.1, includes the requirements for MLPP. In general, the MLPP requirements in this section apply to IP-based Precedence-Based Assured Services (PBAS), i.e., the use of the term "MLPP" in this section is not meant to restrict these requirements to services provided by TDM-based networks. The MLPP service applies to the MLPP service domain only. Connections and resources that belong to a call from an MLPP

subscriber shall be marked with a precedence level and domain identifier (consistent with ANSI Standards T1.619-1992 and T1.619a-1994) and shall be preempted only by calls of higher precedence from MLPP users in the same MLPP service domain. The SUT met this requirement with testing and the vendor's LoC.

(a) The UCR 2013, Errata 1, section 2.25.1.1, states that the SC, SS, PEI and AEI shall provide five precedence levels. The precedence levels listed from lowest to highest are ROUTINE, PRIORITY, IMMEDIATE, FLASH, and FLASH OVERRIDE. The SUT met this requirement with testing and the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.25.1.2, includes the requirements for invocation and operation. The SUT met this requirement with testing and the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.25.1.3, includes the requirements for preemption in the network. The SUT met this requirement with testing and the vendor's LoC with the following minor exception. During testing, the SUT sent a 500-server error when an unanswered trunk call was preempted for reuse. DISA has accepted the vendor's POA&M and adjudicated this discrepancy as minor. During testing, the SUT AudioCodes gateway did not support Method 2 hunt sequence. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to conditional in the next version of the UCR.

(d) The UCR 2013, Errata 1, section 2.25.1.4, includes the requirements for preempt signaling. The SUT met this requirement with testing and the vendor's LoC.

(e) The UCR 2013, Errata 1, section 2.25.1.5, includes the requirements for analog line MLPP. The SUT met this requirement with testing.

(f) The UCR 2013, Errata 1, section 2.25.1.6, includes the requirements for ISDN MLPP BRI. The SUT met this requirement with testing. The SUT does not support ISDN BRI, which is an optional requirement.

(g) The UCR 2013, Errata 1, section 2.25.1.7, includes the requirements for ISDN MLPP PRI. The SUT met this requirement for T1 PRI with testing with the following minor exception. During testing, the SUT allowed unlike service domains to preempt. DISA has accepted the vendor's POA&M and adjudicated this discrepancy as minor. The SUT is certified in the United States, including the CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.

(h) The UCR 2013, Errata 1, section 2.25.1.8, includes the requirements for MLPP interactions with common optional features and services. The SUT met this requirement with testing and the vendor's LoC.

(i) The UCR 2013, Errata 1, section 2.25.1.9, includes the optional requirements for MLPP interactions with Electronic Key Telephone Systems (EKTS) features. The SUT does not support this optional requirement.

(j) The UCR 2013, Errata 1, section 2.25.1.10, states that call gapping shall not apply to FLASH and FLASH OVERRIDE calls. In addition, FLASH and FLASH OVERRIDE calls shall be exempt from Cancel to (CANT) and Cancel from (CANF). The SUT met this requirement with the vendor's LoC.

(2) The UCR 2013, Errata 1, section 2.25.2, includes requirements for Signaling.

(a) The UCR 2013, Errata 1, section 2.25.2.2, states the UC signaling appliance systems shall meet the network power systems requirements specified in the Telcordia Technologies GR-506-CORE, Paragraph 2.1. The SUT met this requirement with the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.25.2.3, includes the requirements for line signaling. The SUT met this requirement with the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.25.2.4, includes the optional requirements for trunk supervisory signaling. The SUT met the optional requirement for reverse battery with the vendor's LoC. The SUT does not support the other optional requirements.

(d) The UCR 2013, Errata 1, section 2.25.2.5, includes the requirements for control signaling. The SUT met this requirement with testing and the vendor's LoC with the following minor exception. During testing, the SUT did not support dial pulse signaling. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to conditional in the next version of the UCR.

(e) The UCR 2013, Errata 1, section 2.25.2.7, includes the requirements for ISDN Digital Subscriber Signaling System No. 1 (DSS1) signaling. The SUT met this requirement with the vendor's LoC.

(3) The UCR 2013, Errata 1, section 2.25.3, includes requirements for ISDN. The UC signaling appliance systems shall provide the ISDN BRI and PRI capabilities shown in Tables 2.25-9 through 2.25-13 as annotated in the tables. The requirements for PRI access, call control, and signaling in Table 2.25-12 are required. The SUT met this requirement with the vendor's LoC.

(4) The UCR 2013, Errata 1, section 2.25.4, includes requirements for backup power. UC shall have backup power to maintain continuous operation whenever the primary source of power is disrupted. Back-up power design and implementation shall be incorporated into the design to assure that UC meets the reliability requirements of UCR 2013, Section 15, Reliability. General power requirements are described in Telcordia Technologies GR-513-CORE. Following the risk avoidance guidance in Telcordia Technologies GR-513-CORE, the backup power design shall minimize the probability of a complete loss of UC appliance system power.

This is a non-testable requirement. The SUT must be installed following this guidance.

(5) The UCR 2013, Errata 1, section 2.25.5, includes Echo Canceller requirements. This section provides the requirements for echo control equipment in the UC network. All MG EC devices are required to meet the requirements. The SUT met this requirement with testing and the vendor's LoC.

(a) The UCR 2013, Errata 1, section 2.25.5.1, includes the requirements for EC functionality. The SUT met this requirement with testing and the vendor's LoC.

(b) The UCR 2013, Errata 1, section 2.25.5.2, includes the requirements for 2100-Hertz EC disabling tone capability. The SUT met this requirement with testing and the vendor's LoC.

(c) The UCR 2013, Errata 1, section 2.25.5.3, states that the EC shall be able to be connected to either analog and/or digital transmission facilities. An analog trunk interface shall be able to provide echo cancellation on a per-trunk basis. A digital trunk interface shall be implemented on a digital basis without conversion to analog. The digital EC shall treat all DS0 channels (PCM-24, PCM-30, or more for SONET) independently. The SUT met this requirement with testing and the vendor's LoC.

(d) The UCR 2013, Errata 1, section 2.25.5.4, includes the requirements for Echo Cancellation on PCM circuits. The SUT met this requirement with testing and vendors LoC.

(e) The UCR 2013, Errata 1, section 2.25.5.5, includes the requirements for device management. The SUT met this requirement with testing.

(f) The UCR 2013, Errata 1, section 2.25.5.6, states that the EC reliability and availability shall conform to Section 5 of Telcordia Technologies GR-512-CORE, as specified for individual devices. The vendor shall provide a reliability model for the system, showing all calculations along with how the overall availability will be met, if requested. The SUT met this requirement with testing.

(6) The UCR 2013, Errata 1, section 2.25.6, includes the requirements for VoIP system latency for MG trunk traffic. The SUT met this requirement with testing.

x. The UCR 2013, Errata 1, section 4, includes the Information Assurance requirements. These requirements are based on Defense Information System Agency (DISA) Facility Security Office (FSO) Security Technical Implementation Guides (STIGs) and Security Requirement Guides (SRGs). The STIGs and SRGs now serve as a primary baseline for purely IA-focused requirements, and this UCR section focuses on those requirements that impact interoperability from an IA standpoint. The SUT met these requirements with the vendor's LoC with the minor exceptions in the following paragraphs. These requirements were removed from the UCR 2013, Change 1 and have immediate applicability. Therefore, there is no operational impact. In addition, Information Assurance is tested by separate IA test teams and the results published in a separate report, Reference (d).

1. During testing, the SUT did not allow users to place emergency calls without authenticating.

2. Per the vendor's LoC, the SUT did not fully support SNMPv3.

3. The SUT Polycom VVX 510 and 610 video phones were interoperable with other tested video endpoints listed on the UC APL except for the Cisco 99xx video EIs. When the Cisco 99xx series video phones originate a video session with the SUT VVX 510 and 610 video phones it results in two-way audio and one-way video (VVX has no video). DISA has accepted the vendor's POA&M and adjudicated this discrepancy as minor.

4. The SUT however, does not send Option Pings but does respond to Option Pings, however this discrepancy did not impact Alternative B failover. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

y. The UCR 2013, Errata 1, section 5, includes the Internet Protocol version 6 requirements. The SC/CCA application in conjunction with the VVoIP EI and Media Gateway (MG) must be IPv6-capable. The guidance in Table 5.2-4 for NA/SS applies. In addition, the requirements throughout section 5 for NA/SS apply. The SUT met the requirements with the vendor's LoC with the exceptions listed in the following subparagraphs. The DoD CIO reviewed and approved the vendor's POA&M and issued a waiver on 28 July 2015 for the following IPv6 requirements:

1. Per the vendor's LoC, the SUT does not fully support RFC 4213.

2. Per the vendor's LoC, the SUT does not fully support RFC 4291.

3. Per the vendor's LoC, the SUT does not support RFC 4007.

4. Per the vendor's LoC, the SUT does not support RFC 4861.

5. Per the vendor's LoC, the SUT's MAS does not support IPv6.

6. Per the vendor's LoC, the Polycom VVX (ROEI) does not support ANAT.

7. Per the vendor's LoC, the SUT does not support RFC 3484.

8. Per the vendor's LoC, the SUT soft client does not support IPsec and RFC 4301.

z. AS-SIP 2013 is the standard signaling protocol used within the Department of Defense (DoD) information systems networks that provide End-to-End Assured Services. This section defines requirements for AS-SIP signaling requirements for operation within an IP-only network infrastructure and for inter-working between IP network segments and legacy Time-Division Multiplexing (TDM) network segments in a hybrid IP and TDM network infrastructure. The

SUT met these requirements with testing and the vendor's LoC with the exceptions listed in the following sub-paragraphs.

1. During testing, the SUT did not allow users to place emergency calls without authenticating. This requirement was removed from the UCR 2013, Change 1 and has immediate applicability. Therefore, there is no operational impact.

2. Per the vendor's LoC, the SUT did not fully support SNMPv3. This requirement was removed from the UCR 2013, Change 1 and has immediate applicability. Therefore, there is no operational impact.

3. During testing, the SUT soft client was unable to establish two-way video with other video systems. DISA has accepted the vendor's POA&M and adjudicated this as critical for video on the soft client. The soft client is certified for audio only.

7. Hardware/Software/Firmware Version Identification: Table 3-3 provides the SUT components' hardware, software, and firmware tested. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic. Table 3-4 provides the hardware, software, and firmware of the components used in the test infrastructure.

8. TESTING LIMITATIONS. None.

9. CONCLUSION(S). The SUT meets the critical interoperability requirements for an LSC in accordance with the UCR and is certified for joint use with other UC Products listed on the APL. The SUT meets the interoperability requirements for the interfaces listed in Table 3-1.

DATA TABLES

Table 3-1. Interface Status

Interface	Applicability (See note 1.)	Status	Remarks
Network Management Interfaces			
10BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface.
100BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface.
1000BaseT	Conditional	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface.
Network Interfaces (Line and Trunk)			
10BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface with the SUT PEIs and softphones.
100BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface with the SUT PEIs and softphones.
1000BaseT	Required	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface with the SUT PEIs and softphones.
2-wire analog (line only)	Required	Met	The SUT met the critical CRs and FRs for the 2-wire analog interface with the SUT 2-wire secure and non-secure analog instruments.
ISDN BRI	Conditional	Not Tested	The SUT does not support this conditional line interface.
Legacy Interfaces (External)			
ISDN T1 PRI (ANSI T1.619a)	Required	Met	The SUT met the critical CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Required	Met	The SUT met the critical CRs/FRs. This interface provides PSTN connectivity.
T1 CCS7 (ANSI T1.619a)	Conditional	Not Tested	The SUT does not support this conditional interface.
T1 CAS	Conditional	Not Tested	Although the SUT supports this conditional interface, it was not tested and is not covered under this certification.
E1 PRI (ITU-T Q.955.3)	Required	Not Tested	The SUT does not support this required interface. This interface provides OCONUS MLPP connectivity in ETSI-compliant countries. (See note 2.)
E1 PRI (ITU-T Q.931)	Required	Not Tested	The SUT does not support this required interface. This interface provides OCONUS connectivity in ETSI-compliant countries. (See note 2.)

NOTES:
1. These high-level requirements refer to a detailed list of requirements provided in Table 3-2.
2. The SUT is certified in the United States, including the CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries..

LEGEND:

10BaseT	10 Mbps Ethernet	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
100BaseT	100 Mbps Ethernet	Mbps	Megabits per second
1000BaseT	1000 Mbps Ethernet	MLPP	Multi-Level Precedence and Preemption
ANSI	American National Standards Institute	NI-2	National ISDN Standard 2
AS-SIP	Assured Services Session Initiation Protocol	OCONUS	Outside the Continental United States
BRI	Basic Rate Interface	PEI	Proprietary End Instrument
CAS	Channel Associated Signaling	PRI	Primary Rate Interface
CCS7	Common Channel Signaling Number 7	PSTN	Public Switched Telephone Network
CONUS	Continental United States	Q.931	Signaling Standard for ISDN
CR	Capability Requirement	Q.955.3	ISDN Signaling Standard for E1 MLPP
DSN	Defense Switched Network	SS7	Signaling System 7
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
ETSI	European Telecommunications Standards Institute	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FR	Functional Requirement	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IEEE	Institute of Electrical and Electronics Engineers		
ISDN	Integrated Services Digital Network		

Table 3-2. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
1	**Voice Features and Capabilities**			
	Call Forwarding	Required	2.2.1	Met
	MLPP Interactions with Call Forwarding	Conditional	2.2.2	Met
	Precedence Call Waiting	Required	2.2.3	Met
	Call Transfer	Required	2.2.4	Met
	Call Hold	Required	2.2.5	Met
	Three-Way Calling	Required	2.2.6	Met
	Hotline Service	Optional	2.2.7	Met
	Calling Number Delivery	Required	2.2.8	Met
	Call Pick-Up	Conditional	2.2.9	Met
	Precedence Call Diversion	Required	2.2.10	Met
	Public Safety Voice Features	Required	2.2.11	Met
2	**Assured Services Admission Control**			
	ASAC Requirements Related to Voice	Required	2.3.1	Met
	ASAC Requirements for the SC and SS Related to Video Services	Required	2.3.3	Met
3	**Signaling Protocols**			
	Signaling Performance Guidelines	Required	2.4	Partially Met (See note 2.)
4	**Registration and Authentication**			
	Registration and Authentication	Required	2.5	Met
5	**SC and SS Failover and Recovery**			
	SC Failover: Alternative A: The SC-Generated OPTIONS Method	Required	2.6.1	Not Tested (See note 3.)
	SC Failover: Alternative B: The SBC-Generated OPTIONS Method	Required	2.6.2	Met
6	**Product Interface**			
	Internal Interface	Required	2.7.1	Met
	External Physical Interfaces Between Network Components	Required	2.7.2	Met
	Interfaces to Other Networks	Required	2.7.3	Met
	DISA VVOIP EMS Interface	Required	2.7.4	Met
7	**Product Physical, Quality, and Environmental Factors**			
	Product Quality Factors	Required	2.8.2	Met
	Voice Service Quality	Required	2.8.4	Met
8	**End Instruments (including tones and announcements)**			
	IP Voice End Instruments	Required	2.9.1	Partially Met (See notes 4,5.)
	Analog and ISDN BRI Telephone Support	Required	2.9.2	Met (See note 6.)
	Video End Instrument	Required	2.9.3	Met
	Authentication to SC	Required	2.9.4	Met
	EI interface to the ASLAN	Required	2.9.5	Met
	Operational Framework for AEIs and Video EIs	Required	2.9.6	Partially Met (See note 7.)
	Multiple Call Appearance Requirements for AS-SIP EIs	Required	2.9.7	Met
	PEIs, AEIs, TAs, and IADs Using the V.150.1 Protocol	Required	2.9.8	Met
	UC Products with Non-Assured Service Features	Conditional	2.9.9	Met
	ROUTINE-only EIs	Required	2.9.10	Met

Table 3-2. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
9	**Session Controller**			
	PBAS/ASAC	Required	2.10.1	Met
	SC Signaling	Required	2.10.2	Met
	Session Controller Location Service	Required	2.10.3	Met
	SC Management Function	Required	2.10.4	Met
	SC-to-VVoIP EMS Interface	Required	2.10.5	Met
	SC Transport Interface Functions	Required	2.10.6	Met
	Custom Line-Side Features Interference	Conditional	2.10.7	Not tested
	Loop Avoidance for SCs	Required	2.10.8	Met
	Local Session Controller Application	Required	2.10.9	Met
10	**AS-SIP Gateways**			
	AS-SIP TDM Gateway	Conditional	2.11.1	Met
	AS-SIP IP Gateway	Conditional	2.11.2	Not Tested
	AS-SIP – H.323 Gateway	Conditional	2.11.3	Not Tested
11	**Call Connection Agent**			
	Introduction	Required	2.14.1	Met
	Functional	Required	2.14.2	Met
	CCA-IWF Signaling Protocol Support	Required	2.14.4	Met
12	**CCA Interaction with Network Appliances and Functions**			
	CCA Interactions with Transport Interface Functions	Required	2.15.1	Met
	CCA Interactions with the SBC	Required	2.15.2	Met
	CCA Support for Admission Control	Required	2.15.3	Met
	CCA Support for User Features and Services	Required	2.15.4	Met
	CCA Support for Information Assurance	Required	2.15.5	Met
	CCA Interactions with End Instruments	Required	2.15.8	Met
	CCA Support for Assured Services Voice and Video	Required	2.15.9	Met
	CCA Interactions with Service Control Functions	Required	2.15.10	Met
13	**Media Gateway**			
	MG Call Denial Treatments to Support CAC	Required	2.16.1	Met
	MG Interfaces to TDM NEs in DoD Networks: PBXs, EOs, and MFSs	Required	2.16.2	Met
	MG Interfaces to TDM NEs in Allied and Coalition Partner Networks	Required	2.16.3	Partially Met (See note 2.)
	MG Interfaces to TDM NEs in the PSTN in the United States	Required	2.16.4	Met
	MG Interfaces to TDM NEs in OCONUS PSTN Networks	Required	2.16.5	Met
	MG Support for ISDN PRI Trunks	Required	2.16.6	Met
	MG Support for CAS Trunks	Optional	2.16.7	Met
	MG Requirements: VoIP Interfaces Internal to an Appliance	Required	2.16.8	Met
	MG Requirements for Echo Cancellation	Required	2.16.9	Met
	MG Requirements for Clock Timing	Required	2.16.10	Met
	MGC-MG CCA Functions	Required	2.16.11	Met
	MGs Using the V.150.1 Protocol	Conditional	2.16.12	Partially Met (See note 7.)
	Remote Media Gateway	Conditional	2.16.13	Not Tested
14	**Worldwide Numbering & Dialing Plan**			
	Worldwide Numbering and Dialing Plan	Required	2.18.1	Partially Met (See notes 8, 9.)

Table 3-2. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/Function	Applicability (See note 1.)	UCR Reference	Status
15	**Management of Network Devices**			
	General Management	Required	2.19.1	Met
	Requirements for FCAPS Management	Required	2.19.2	Met
16	**V.150.1 Modem Relay Secure Phone Support**			
	V.150.1 Modem Relay Secure Phone Support	Required	2.20	Partially Met (See note 7.)
17	**Requirements for Supporting AS-SIP Based Ethernet Devices for Voicemail Systems**			
	Requirements for Supporting AS-SIP Message Waiting Indications on AS-SIP EIs, TAs, and IADs	Optional	2.21.1	Met
18	**Local Attendant Console Features**			
	Local Attendant Console Features	Optional	2.22	Not Tested
19	**MSC and SSC**			
	MSC and SSC	Optional	2.23	Not Tested
20	**MSC, SSC, AND Dynamic ASAC Requirements in Support of Bandwidth-constrained Links**			
	MSC and SSC Architecture	Optional	2.24.1	Not Tested
	Dynamic ASAC	Optional	2.24.2	Not Tested
21	**Other UC Voice**			
	Multilevel Precedence and Preemption	Required	2.25.1	Partially Met (See notes 10, 11, 12, 13.)
	Signaling	Required	2.25.2	Partially Met (See note 14.)
	ISDN	Required	2.25.3	Met
	Backup Power	Required	2.25.4	Met
	Echo Canceller	Required	2.25.5	Met
	VoIP System Latency for MG Trunk Traffic	Required	2.25.6	Met
22	**IA Requirements**			
	Security Requirements	Required	4.2	Partially Met (See notes 15, 16.)
23	**IPv6 Requirements**			
	IPv6	Required	5.2	Not Tested (See note 17.)
24	**Assured Services - Session Initiation Protocol (AS-SIP 2013)**			
	AS-SIP	Required	AS-SIP	Partially Met (See note 18.)

Table 3-2. SUT Capability Requirements and Functional Requirements Status (continued)

NOTES:

1. The annotation of ‘required’ refers to a high-level requirement category. The applicability of each sub-requirement is provided in the UCR, Reference (b). The system under test does not need to provide conditional requirements. However, if a capability is provided, it must function according to the specified requirements.
2. Although the SUT supports E1 interfaces, they were not tested and are not covered under this certification. This discrepancy was adjudicated by DISA as minor with vendor POA&M and the following condition of fielding: The SUT is not certified for joint use outside CONUS in ETSI-compliant countries.
3. The SUT does not support this failover method. The SUT supports Alternative B for failover.
4. The SUT Polycom VVX 510 and 610 video phones were interoperable with other tested video endpoints listed on the UC APL except for the Cisco 99xx video EIs. When the Cisco 99xx series video phones originate a video session with the SUT VVX 510 and 610 video phones it results in two-way audio and one-way video (VVX has no video). DISA has accepted the vendor’s POA&M and adjudicated this discrepancy as minor.
5. During testing, the SUT soft client was unable to establish two-way video with other video systems. DISA has accepted the vendor’s POA&M and adjudicated this as critical for video on the soft client. The soft client is certified for audio only.
6. The SUT does not support ISDN BRI.
7. The Polycom 310, 410, 510, and 610 EIs do not divert precedence above ROUTINE calls to an attendant or Alternate DN. DISA has adjudicated this discrepancy as minor. This requirement was changed in the UCR 2013, Change 1 to allow precedence calls above ROUTINE to be answered by an ROEI if resources are available.
8. During testing, the AudioCodes MG3K and M800 could not dynamically invoke VBD G.711 or V.150.1. The SUT supports V.150.1 however when G.711 Voice Codec is negotiated the SUT fails to support bi-directional secure calls in pass-through mode. The SUT DISA has accepted the vendor’s POA&M and adjudicated this as minor with the Condition of Fielding that the SUT cannot operate with V.150 until this discrepancy is corrected.
9. Per the vendor’s LoC, TEO AEIs do not support integration with directory. DISA has adjudicated this discrepancy as minor.
10. During testing, the SUT did not support Domain Directory per the requirement. DISA has accepted the vendor’s POA&M and adjudicated this discrepancy as minor.
11. During testing, the SUT allowed unlike service domains to preempt. DISA has accepted the vendor’s POA&M and adjudicated this discrepancy as minor.
12. During testing, the SUT sent a 500-server error when an unanswered trunk call was preempted for reuse. DISA has accepted the vendor’s POA&M and adjudicated this discrepancy as minor.
13. During testing, the SUT AudioCodes gateway did not support Method 2 hunt sequence. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to conditional in the next version of the UCR.
14. During testing, the SUT sent a 500-server error when an unanswered trunk call was preempted for reuse. DISA has accepted the vendor’s POA&M and adjudicated this discrepancy as minor.
15. During testing, the SUT did not support dial pulse signaling. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to conditional in the next version of the UCR.
16. All IA requirements in UCR, section 4, have been changed in UCR 2013, Change 1, with the following caveat: The requirements in section 4 will not be evaluated in interoperability test plans and are the responsibility of cyber security testing with the intent to minimize redundancy in cyber security test procedures and reports. The update in UCR 2013, Change 1 has immediate applicability.
    - During testing, the SUT did not allow users to place emergency calls without authenticating TLS.
    - Per the vendor’s LoC, the SUT did not fully support SNMPv3.
17. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (d).
18. Per the vendor’s LoC, the SUT does not fully support IPv6. The DoD CIO waived all of the IPv6 discrepancies noted in this certification letter on 28 July 2015 with vendor’s POA&M. Therefore, IPV6 was not tested and is not included in this certification.
    - Per the vendor’s LoC, the SUT does not fully support RFC 4213.
    - Per the vendor’s LoC, the SUT does not fully support RFC 4291.
    - Per the vendor’s LoC, the SUT does not support RFC 4007.
    - Per the vendor’s LoC, the SUT does not support RFC 4861.
    - Per the vendor’s LoC, the SUT’s MAS does not support IPv6.
    - Per the vendor’s LoC, the Polycom VVX (ROEI) does not support ANAT.
    - Per the vendor’s LoC, the SUT does not support RFC 3484.
    - Per the vendor’s LoC, the SUT soft client does not support IPsec and RFC 4301.

Table 3-2. SUT Capability Requirements and Functional Requirements Status (continued)

LEGEND:

AEI	AS-SIP End Instrument	MAS	Media Application Server
ASAC	Assured Services Admission Control	MFS	Multifunction Switch
ASLAN	Assures Services Local Area Network	MG	Media Gateway
AS-SIP	Assured Services Session Initiation Protocol	MGC	Media Gateway Controller
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
CAC	Common Access Card	MSC	Master Session Controller
CAS	Common Access Signaling	NE	Network Element
CCA	Call Connection Agent	OCONUS	Outside the Continental United States
CIO	Chief Information Officer	PBAS	Precedence Based Assured Services
CONUS	Continental United States	PBX	Private Branch Exchange
CR	Capability Requirement	PEI	Proprietary End Instrument
DISA	Defense Information System Agency	POA&M	Plan of Action and Milestones
DN	Directory Number	PRI	Primary Rate Interface
DoD	Department of Defense	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate	RFC	Request For Change
EI	End Instrument	ROEI	Routine Only End Instrument
EMS	Element Management System	SBC	Session Border Controller
ETSI	European Telecommunications Standards Institute	SC	Session Controller
EO	End Office	SNMPv3	Simple Network Management Protocol version 3
FCAPS	Fault, Configuration, Accounting, Performance, and Security	SS	SoftSwitch
		SSC	Subtended Session Controller
FR	Functional Requirement	SUT	System Under Test
IA	Information Assurance	TA	Terminal Adapter
IAD	Integrated Access Device	TDM	Time Division Multiplexing
ID	Identification	UC	Unified Capabilities
IP	Internet Protocol	UCR	Unified Capabilities Requirements
IPv6	Internet Protocol version 6	VBD	Voice Band Data
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
IWF	Interworking Function	VVoIP	Voice and Video over Internet Protocol
LoC	Letter of Compliance	VVX	Voice and Video eXchange

Table 3-3. SUT Hardware/Software/Firmware Version Identification

Component (See note 1.)	Release	Sub-component (See note 1.)	Function
EXPERiUS Application Server	11.2	Personal Agent (VMWare) (x2) MCP 17.0, Red Hat Linux 6.6	Provides management function for passwords for EXPERiUS user administered passwords, reservation-less conferencing bridges, and unified messaging.
		Session Manager (VMWare) (x2) MCP 17.0, Red Hat Linux 6.6	Provides BBUA, CPL Interpreter, ASAC Budgeting, and address resolution and routing capabilities
		Media Application Server (MAS) (X2) MAS 16.0	Provides flexible high availability media services run time base and packaged with Ad hoc Conferencing, Meet Me Conferencing (audio & video), Tones/Announcements, Music on Hold, Voice Mail
AudioCodes Mediant 3000 (M3000) Gateway	Mediant 6.60A.305.001	Not Applicable	Provides PSTN/DSN capability with high density, carrier grade, trunk gateway for supporting commercial and DSN trunks
	pSoS 2.5.4		
AudioCodes M800 Gateway	Mediant 6.60A.305.001	Not Applicable	Supports commercial and DSN trunks, as well as analog lines with AS-SIP capabilities
	pSoS 2.5.4		
TEO **7810**, **4104** 4101 AS-SIP and SIP telephones	xx.04.22 (See note 2.)	Not Applicable	IP Phone, 10/100/1000 Mbps Ethernet
Polycom VVX 310, VVX 410, VVX 500, VVX 600	5.4.2.0334	Not Applicable	High performance unified communications which demonstrates interoperability with third party end instruments as a VVoIP
Dell Laptop (Management Workstation) (Site-provided Dell Laptop)	Windows 7 SP 1	Not Applicable	Management Workstation
GENBAND Multimedia Client (See note 3.) (Site-provided Dell Laptop)	Windows 7 SP 1	Not Applicable	Soft Client
	GENCom 10.4 v 10.4.1368		
	Axway Desktop Validator 4.11.2.753		
	ActivClient 6.2.0.50		

NOTES:
1. Components bolded and underlined were tested by JITC. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes.
2. The TEO IP 7810, 4104, and 4101 phone units were tested using version xx.04.22 from the previous xx.04.21.
3. The SUT Soft Client is certified for voice only.

LEGEND:

ASAC	Assured Services Admission Control
AS-SIP	Assured Services Session Initiation Protocol
BBUA	Back-to-Back User Agent
CPL	Call Processing Language
DSN	Defense Switched Network
IP	Internet Protocol
JITC	Joint Interoperability Test Command
MAS	Media Application Server
Mbps	Megabits per second
MCP	Media Communications Processor
PSTN	Public Switched Telephone Network
SP	Service Package
SUT	System Under Test
VVoIP	Voice and Video over IP
VVX	Voice and Video eXchange

Table 3-4. Test Infrastructure Hardware/Software/Firmware Version Identification

System Name	Software Release	Function
Required Ancillary Equipment (site-provided)		
Active Directory		
Public Key Infrastructure		
Syslog Server		
Test Network Components		
Avaya CS2100	SE 09.1	MFS
Avaya S8800	CM 6.3.111.0	LSC/SMEO
Cisco UCM	ESC15 Version 10.5	ESC/LSC
Avaya Aura® AS5300	3.0, SP 11	SS/LSC
REDCOM HDX	4.0AR3P9	LSC
REDCOM SLICE	4.0AR3P9	LSC
Siemens EWSD	19d Patch 46	MFS
NEC UNIVERGE 3C	8.5.4.19	LSC
AcmePacket 3820	6.4.1 MR1 Build 14	SBC
AcmePacket 4500	6.4.1 MR1 Build 14	SBC
AcmePacket 3820	6.4.1 MR1 Build 14	SBC
Polycom RMX Video	8.4.2	UCCS
Polycom Real Presence Group Series 500	4.3.2	Video AS-SIP PEI / VTC endpoint
Vidyo Conferencing Suite	3.2.4	Video Only UCCS
Cisco UCM SX Series	CE 8.1	Video
Cisco UCM DX Series	10.2.4.2	VVoIP
Cisco 88XX Series	10.3.2	VVoIP
Cisco 99XX Series	9-4-2SR2-2	VVoIP
Cisco Jabber	8.6.1.0	Soft Client VVoIP
Avaya Aura AS5300	8.1.5196	Soft Client VVoIP
Acano Video	1.8.3	Video
General Dynamics vIPer	5.01	DSCD
Omni Secure Wireline Terminal	B006000010000	DSCD
L3 Communications STE	2.8.1	DSCD
Abacus Call Loader	5.2	Call Loader
Anue Network Emulator	6.1	Network Impairment Device

LEGEND:

AS	Application Server
AS-SIP	Assured Services Session Initiation Protocol
CM	Communication Manager
CS	Communication Server
DSCD	Department of Defense Secure Communications Device
ESC	Enterprise Session Controller
EWSD	Elektronisches Wählsystem Digital
HDX	High Density Exchange
IP	Internet Protocol
LSC	Local Session Controller
MFS	Multifunction Switch
MR1	Maintenance Release 1
PEI	Proprietary End Instrument
SBC	Session Border Controller
SE	Succession Enterprise
SMEO	Small End Office
SP	Service Pack
SS	SoftSwitch
STE	Secure Terminal Equipment
UCCS	Unified Capabilities Conferencing System
UCM	Unified Communications Manager
VVoIP	Voice and Video over IP
VTC	Video Teleconference
XMPP	Extensible Messaging and Presence Protocol

Joint Interoperability Certification Revision History

Revision	Date	Approved By	Comments
NA	10 February 2016	Soamva Duong	This is the original Joint Interoperability Certification Extension.
1	11May 2016	Cary Hogan	The following change was made to the certification: • Memo, Page 5, Table 4: The following components were changed: - TEO IP phone units7810, 4101 and 4104 were changed to reflect the correct software update from 06.04.22.41 to xx.04.22. - A note was added to the LEGEND to reflect the change stating, "The TEO IP 7810, 4104, and 4101 phone units were tested using version xx.04.22 from the previous xx.04.21". • Enclosure 3, Page 3-7, Table 3-3: The following components were changed: - TEO IP phone units7810, 4101 and 4104 were changed to reflect the correct software update from 06.04.22.41 to xx.04.22. - A note was added to the LEGEND to reflect the change stating, "The TEO IP 7810, 4104, and 4101 phone units were tested using version xx.04.22 from the previous xx.04.21".

LEGEND:
IP Internet Protocol
NA Not Applicable
SUT System Under Test